Front-End Electronics based on Waveform Sampling

Feature Extraction

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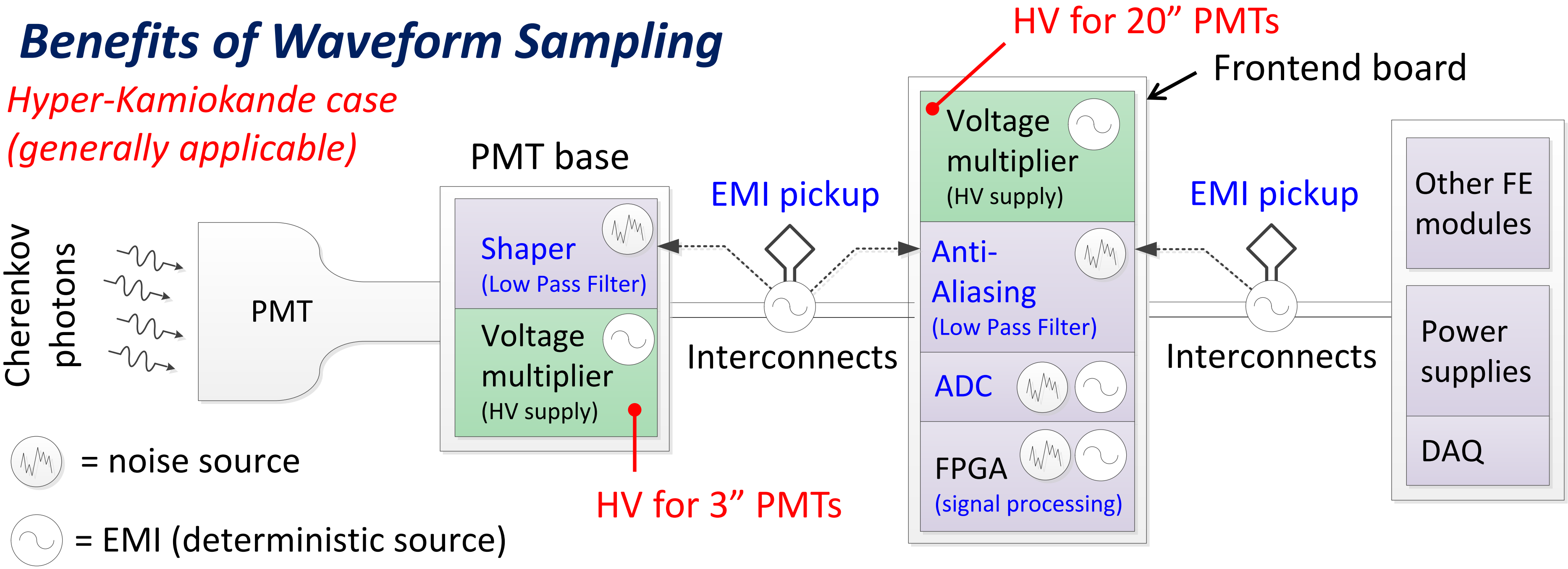
Introduction

Benefits of Waveform Sampling

Hyper-Kamiokande case
(generally applicable)

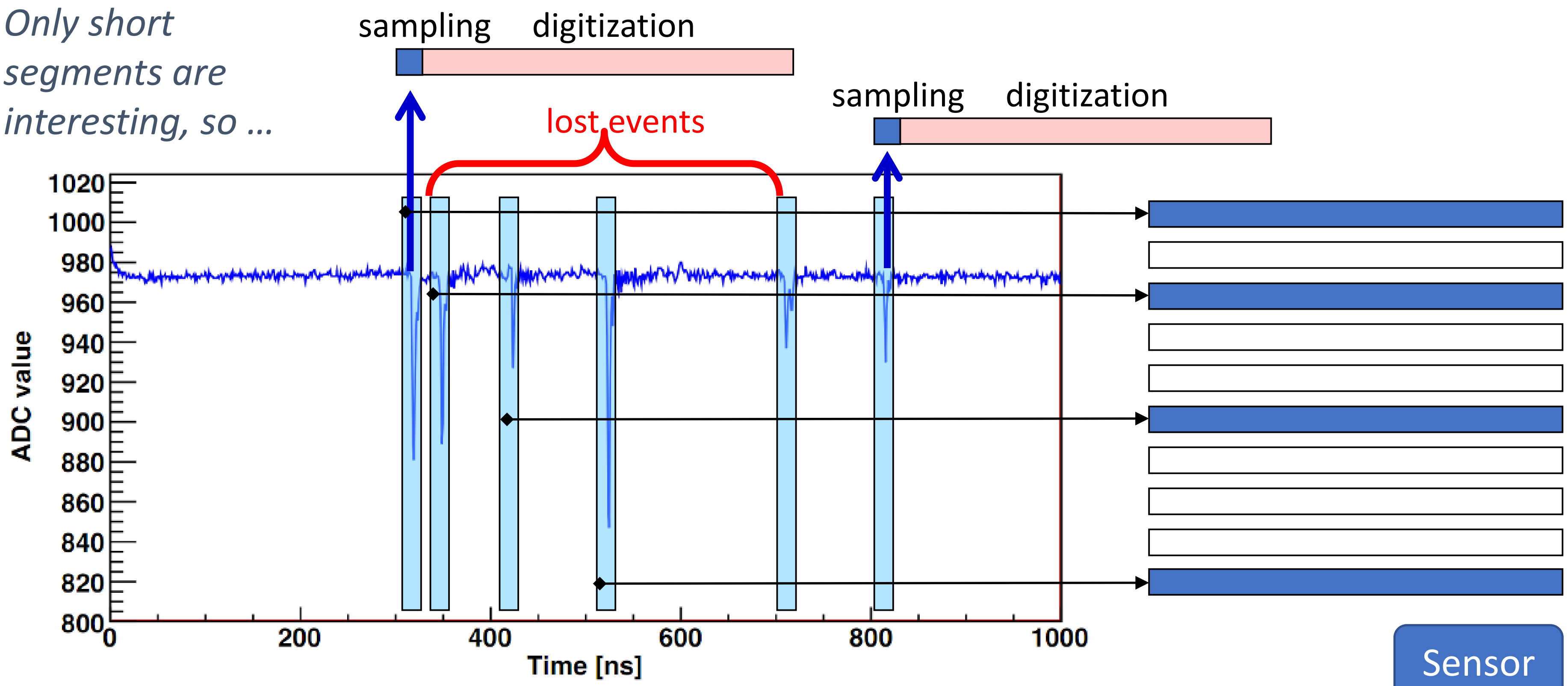


- Possibility to implement completely dead-time free system.
- Ability to disentangle overlapping pulses (pile-up)
- **Can subtract off periodic EMI by digital filters implemented in FPGA firmware.**
- There is a price to pay: **power consumption,** cost, **data rate**.
  - Can we reduce the above without affecting the physics performance?

Fast Digitizer at Reasonable Power & Cost

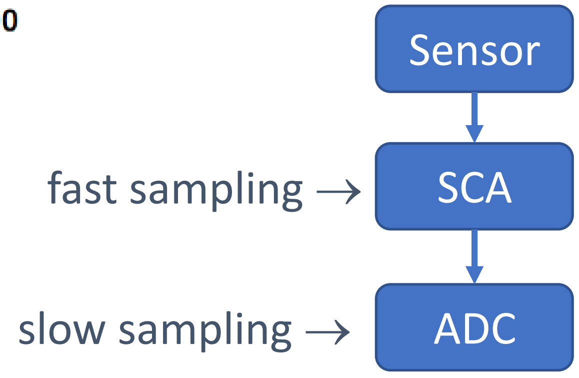
Switched Capacitor Arrays (DRS4 example)

Only short segments are interesting, so …



INTRODUCTION OF DEAD TIME

→ Not a problem if mean inter-pulse period is large compared to the dead time



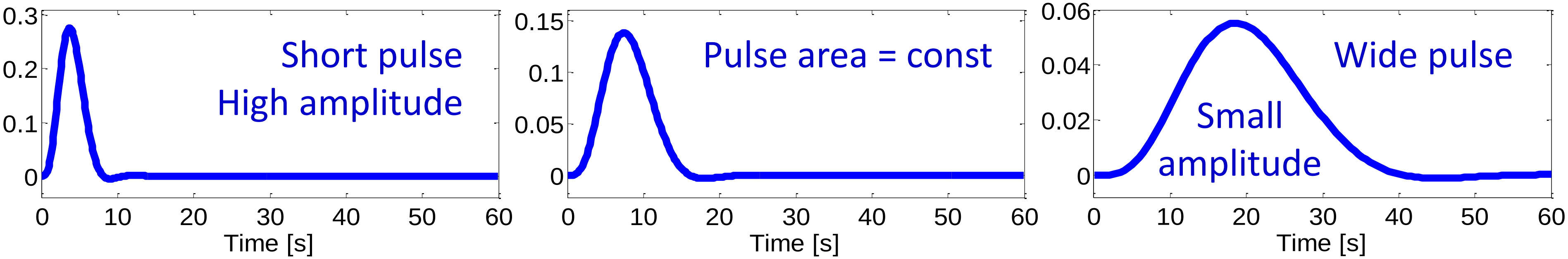
Avoiding dead time in capacitor arrays:

- Use multiple arrays for single waveform
- Use chip with segmented memory (if available)

Study of Sampling Systems

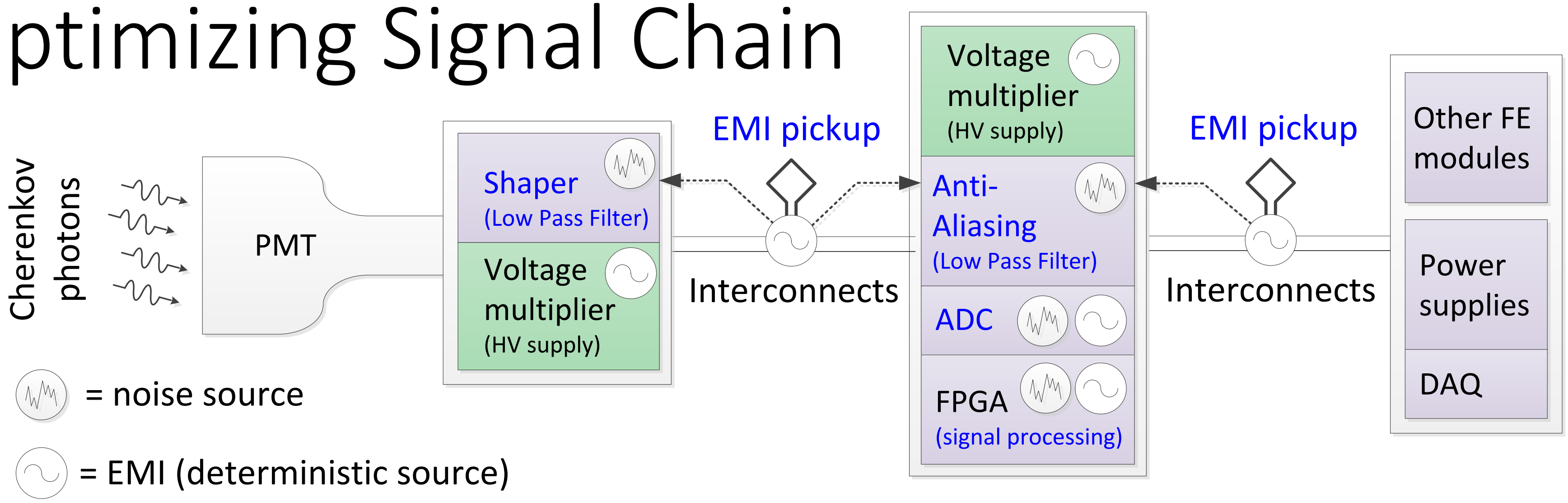
High resolution → Low resolution

How **poor** can the **system specs** be to still be able to tell **when** and how big the **pulse was** with **satisfactory precision**?



High bandwidth → Low bandwidth

Optimizing Signal Chain

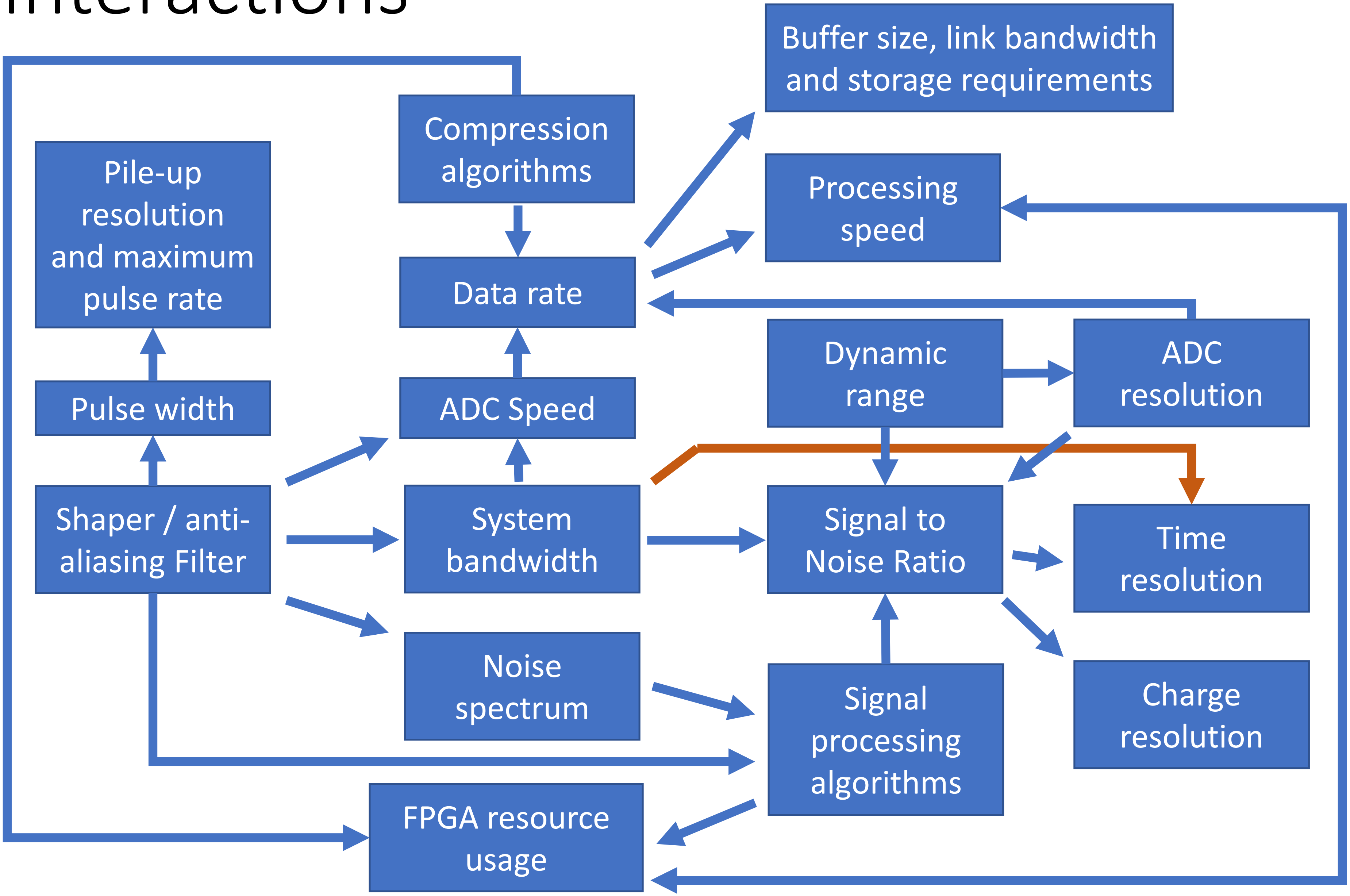


QUESTIONS:

- Type and cutoff frequency of analog shaper/anti-aliasing filter?
- Speed and resolution of the ADC?
- Signal processing methods and sharing of signal processing between FPGA and DAQ
- Optimization of resource usage within the FPGA
- Quality of time & charge estimates
- Two independent compression methods:
  - Waveform (potentially lossy)
  - Time/charge (lossless)
- Disentanglement of pulse pile-up

Need decent model of the full signal chain → having one allows exploration of various variants of shaper/ADC combinations without the need for building prototypes (thus saves labor time)

Interactions

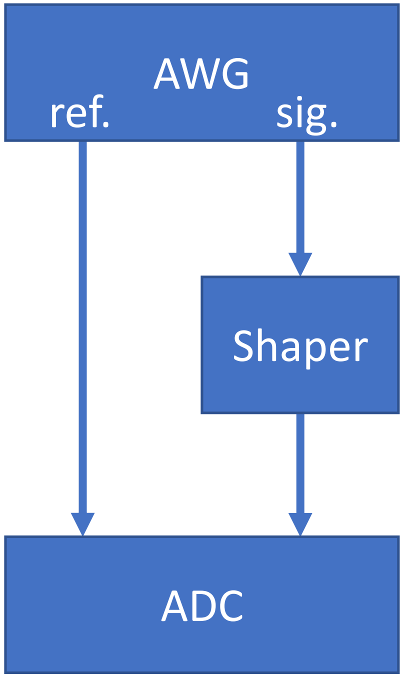


Timing Resolution of Sampling Digitizers

PURPOSE OF THE STUDY:

Determine how fast and how precise does a system needs to be to achieve given performance specs?

- Use AWG instead of PMT.
- Use large reference pulse (timing accuracy σ ≈ 10 ps) and small, shaped signal pulse (1 mV ~ 100 mV).
- Apply signal processing methods and calculate time difference Δt between ref. and sig. channels.
- Repeat multiple times and compute RMS of Δt values.
- Two shapers:
  - 15 ns and 30 ns rise time (10% to 90%), 5-th order **Bessel-type** low-pass filters.
- Shared project WUT/TRIUMF



Agilent 33600A (1 GSPS/80 MHz)

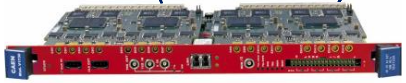
Custom shapers

Commercial ADCs (CAEN)

DT5724 (100 MSPS/14b)

V1720 (250 MSPS/12b)

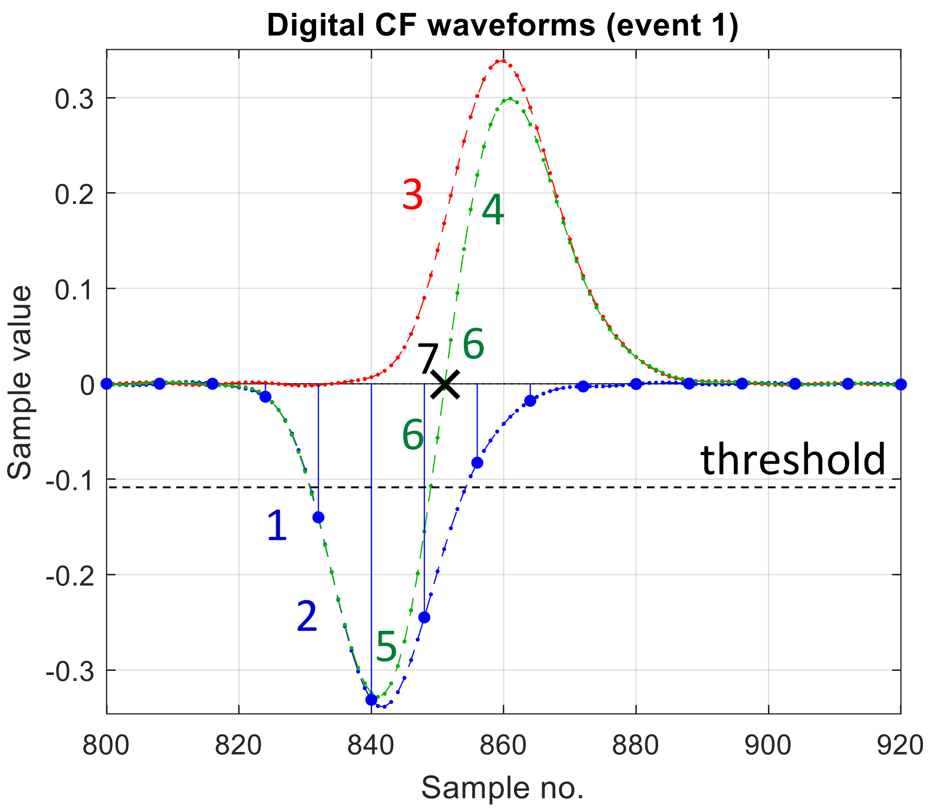
V1730 (500 MSPS/14b)



Signal Processing Methods

Digital Constant Fraction Discriminator:

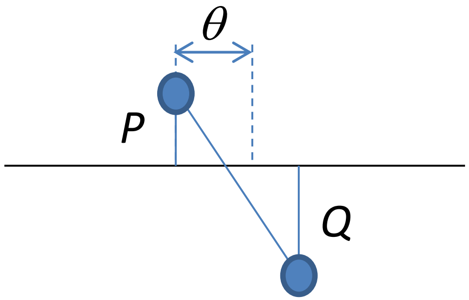
- Simple processing → needs little FPGA resources
- Does not make any assumption as to the pulse shape
- Favors high sampling rate, but some improvements are possible for low sampling rates if pulse shape is invariant
- Poor performance in low SNR conditions



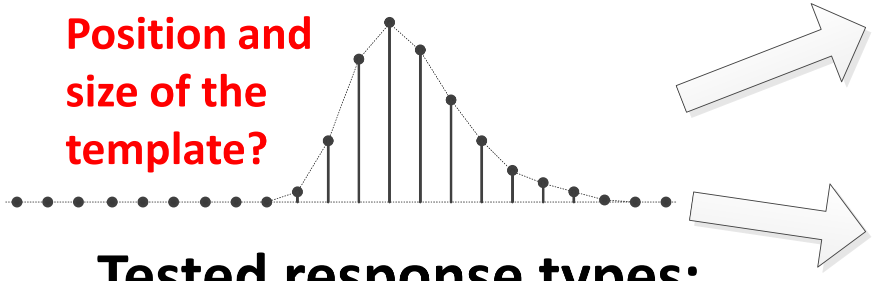
Time errors and possible correction

θ - actual sub-sample shift

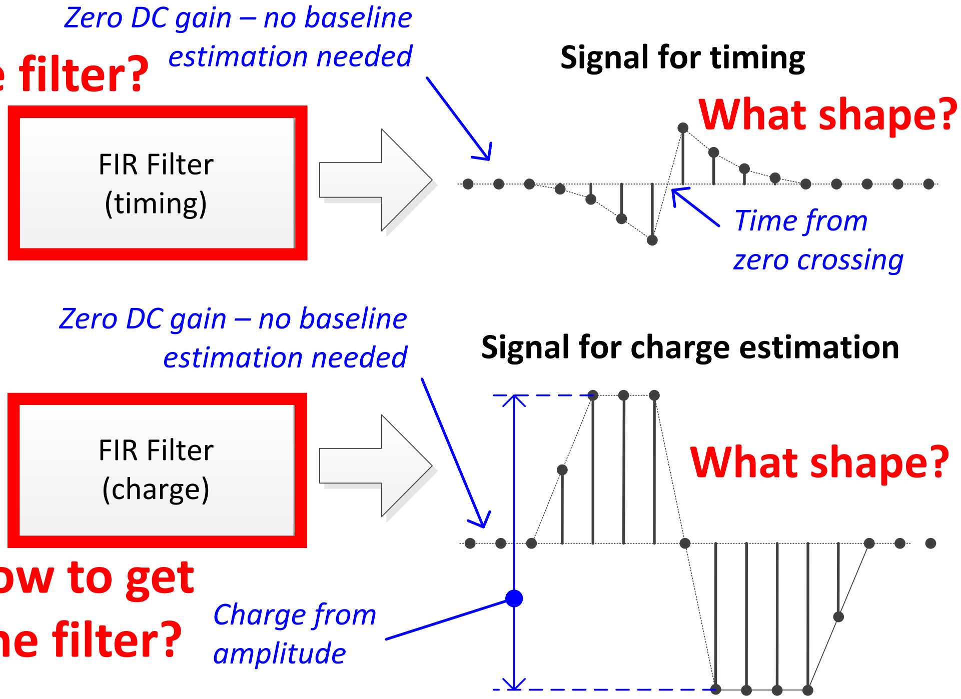
$$CR = \frac{P}{P-Q}$$



Signal Processing – FIR DPLMS



Position and size of the template?



How to get the filter?

Zero DC gain – no baseline estimation needed

What shape?

Time from zero crossing

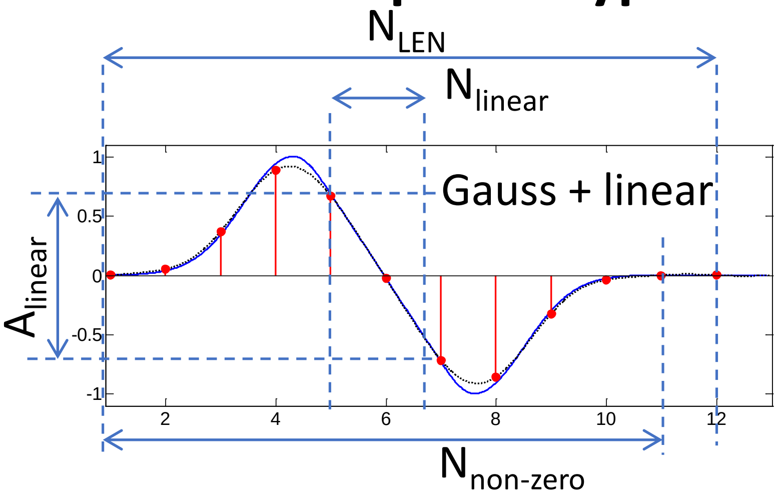
Zero DC gain – no baseline estimation needed

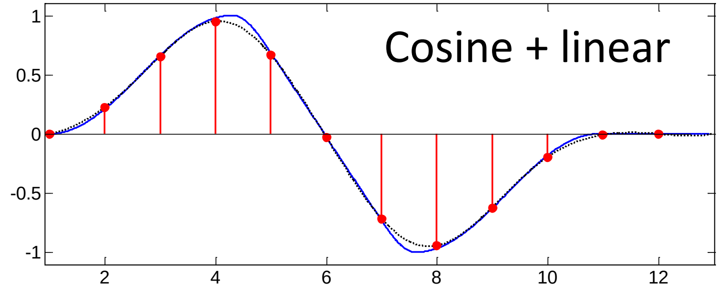
What shape?

How to get the filter?

Charge from amplitude

Tested response types:





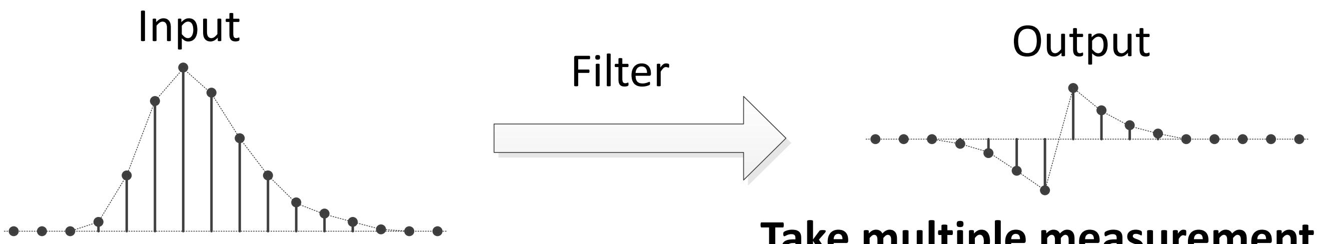
… or simply subtract pedestal and integrate.

- FIR = Finite Impulse Response
- ‘Black-box’ approach → transform **known** input into desired output, don’t care how.
- Arbitrary filter characteristic possible.
- Filter should be ‘optimal’ → **minimize certain cost function (constrained optimization)**.

Gatti E., et al., “*Digital Penalized LMS method for filter synthesis with arbitrary constraints and noise*”, NIM A523, 167-185, 2004

Synthesizing FIR filter – Method 1

Digital Penalized LMS Method



input signal *noiseless signal (our template)* *stationary noise*

$$x[n] = x'[n] + x"[n]$$

number of filter taps *impulse response of the filter*

Filter is **linear**, so the output signal is:

$$y[n] = \sum_{l=0}^{N-1} h[l] \cdot x'[n-l] + \sum_{l=0}^{N-1} h[l] \cdot x"[n-l]$$

Therefore, we can deal with noise and signal components separately

Take multiple measurements, then:

Minimize overall variance of the response:

Sought filter

$$Var(y) = \boldsymbol{h}^{1,N} \cdot \boldsymbol{R}^{N,N} \cdot \boldsymbol{h}^{N,1}$$

Noise auto-covariance matrix

Minimize difference between filter response and our desired response

$$(E(y[k] - v_k))^2 = (\boldsymbol{h}^{1,N} \cdot \boldsymbol{x}'(k)^{N,1} - v_k)^2$$

Value of k-th sample of the response to x'

N past samples of x', starting from k

Gatti E., et al., "*Digital Penalized LMS method for filter synthesis with arbitrary constraints and noise*", NIM A523, 167-185, 2004

Synthesizing FIR filter – Method 1 (cont.)

Digital Penalized LMS Method

Add additional constraints for frequency response, including gain at DC ...

Add constraints related to bit-gain (i.e. how well we are supposed to reject quantization noise) ...

Finally, build the error functional and minimize it:

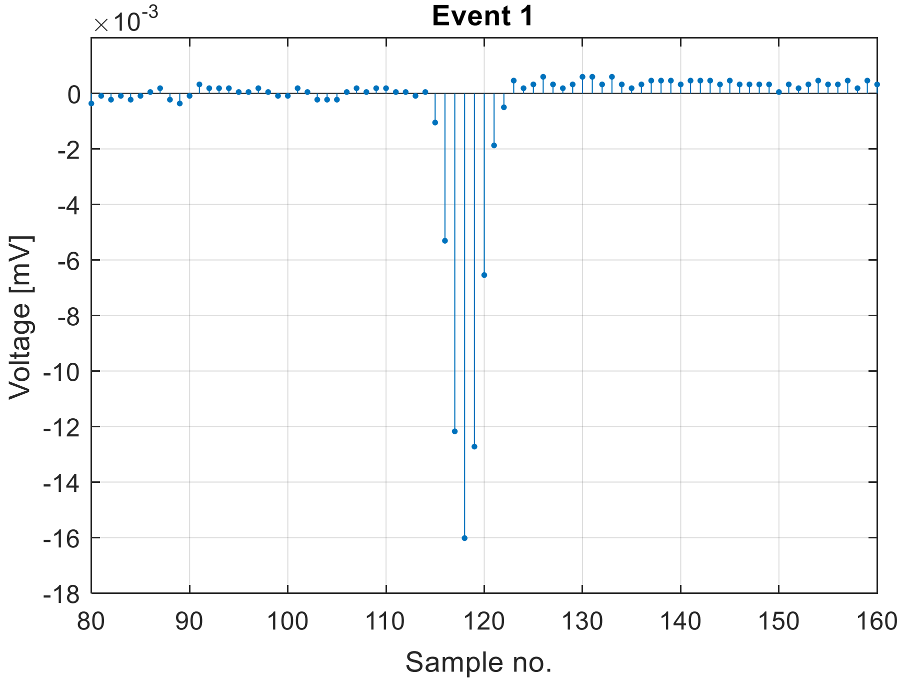
$$Area(FIR) = \frac{Area(y)}{Area(x)}$$

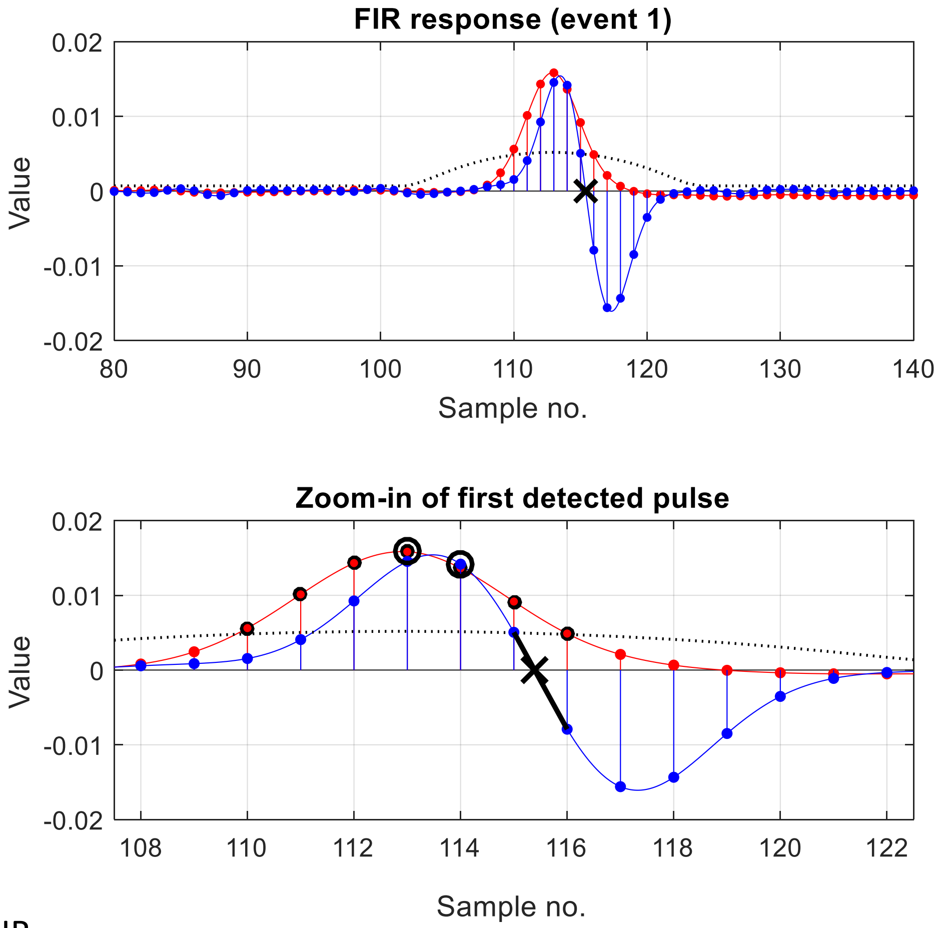
$$\varepsilon^2 = \overbrace{Var(y)}^{\text{Constraint for variance}} + \overbrace{\sum_{k=1}^{N} \alpha_k \cdot (E(y[k]) - v_k)^2}^{\text{Constraints for shape of response to pulse template}} + \overbrace{\sum_{l=1}^{L} \beta_l \cdot \left(|F\{\boldsymbol{h}\}|_{\omega_l}\right)^2}^{\text{Frequency constraints}}$$

$$+ \underbrace{\varphi \cdot (F\{\boldsymbol{h}\}_{\omega=0} - Area(FIR))^2}_{\text{DC gain (i.e. area) constraint}} + \underbrace{\gamma \cdot \sum_{n} (h[n])^2}_{\text{Bit-gain constraint}}$$

All components are square functions, so there exists a global minimum – just need to properly choose ***N***, $\vec{v}$, $\vec{\alpha}$, $\vec{\beta}$, φ and γ → papers don't say much about that

Signal Processing - FIR Filters





- Trigger on matched filter response (red)
- Use adaptive threshold to prevent false positives (dotted black line)
  - Average signal to get the threshold and delay FIR processing to check for pulses and their timing
- Get time using the ‘timing’ filter (blue)
- Apply correction to counteract non-linear shape of the waveform near zero-crossing.

Method assumes that shape is constant

Need on-line Quality Factor to judge accuracy of estimation

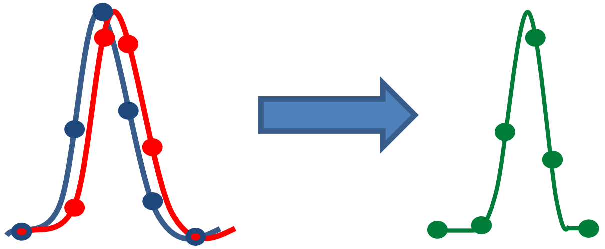
Signal Processing – Continued

Matched FIR Filter and Cross-Correlation Processing:

- Much more complex processing
  - Works well with filter orders of 9-12
- **Assumes that shape is constant**
- Similar timing performance to zero-average FIR filter
- Relatively easy to disentangle piled-up pulses

Misaligned pulses

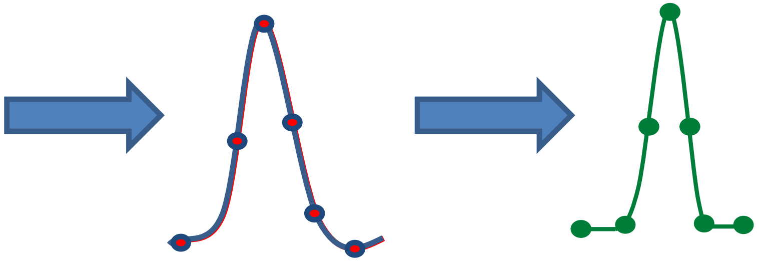
Pulses Cross-correlation



Aligned pulses

Pulses Cross-correlation

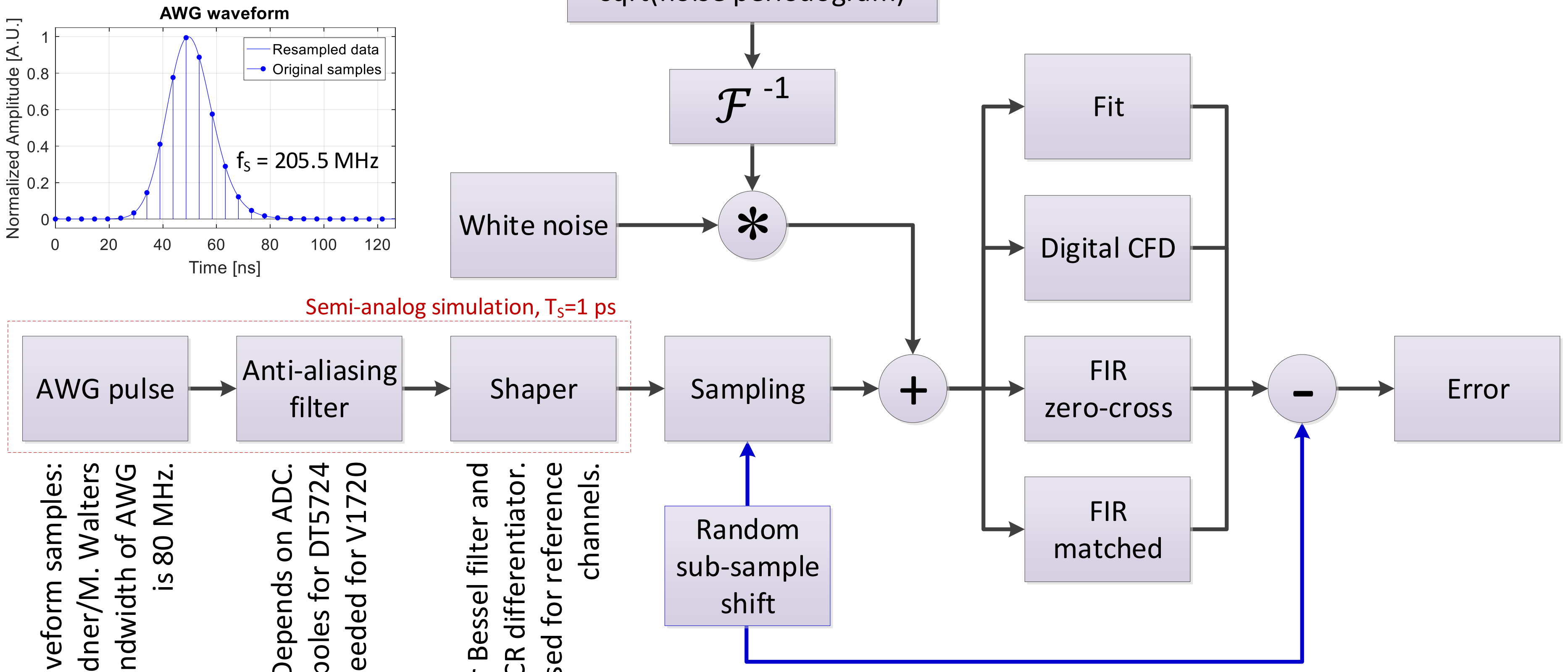
Sub-sample shifts done using windowed sinc interpolation (Blackman window). FFT interpolation also possible if shifting impulse response.



System Model (each channel)

Used 250 MHz data to determine actual AWG f_S

$$\sigma_{final} = \sqrt{{\sigma_{ref}}^2 + {\sigma_{sig}}^2}$$



Waveform samples:
T. Lindner/M. Walters
Analog bandwidth of AWG is 80 MHz.

Depends on ADC.
2 real poles for DT5724
Not needed for V1720

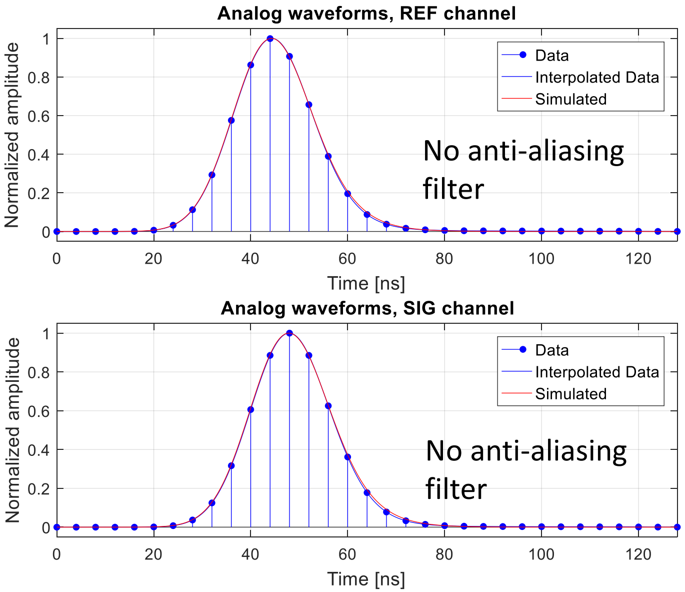
5-th order Bessel filter and CR differentiator.
Not used for reference channels.

All transfer functions (*TF*) calculated in s-domain, then used $\mathcal{L}^{-1}$ to calculate impulse response.

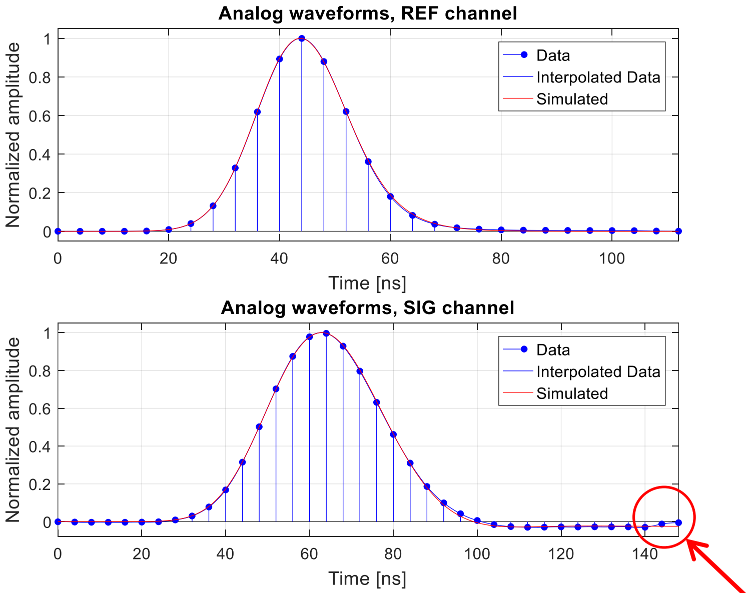
Signal Models

All pulses matched by FWHM

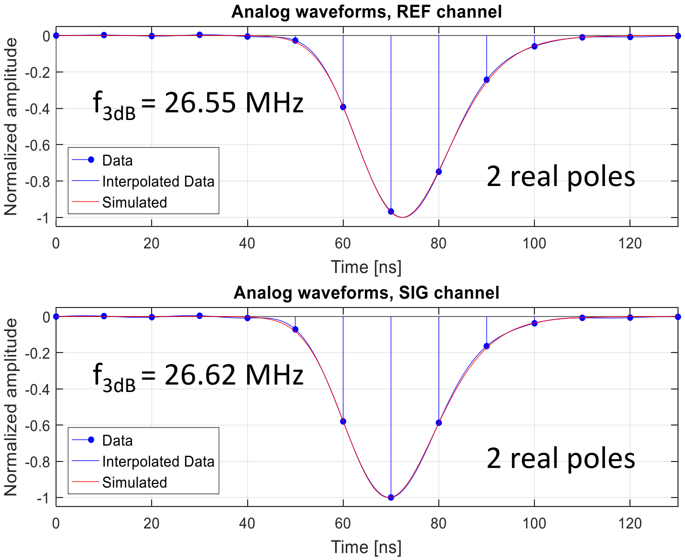
250 MHz,
CH1 = ref, CH2 = ref



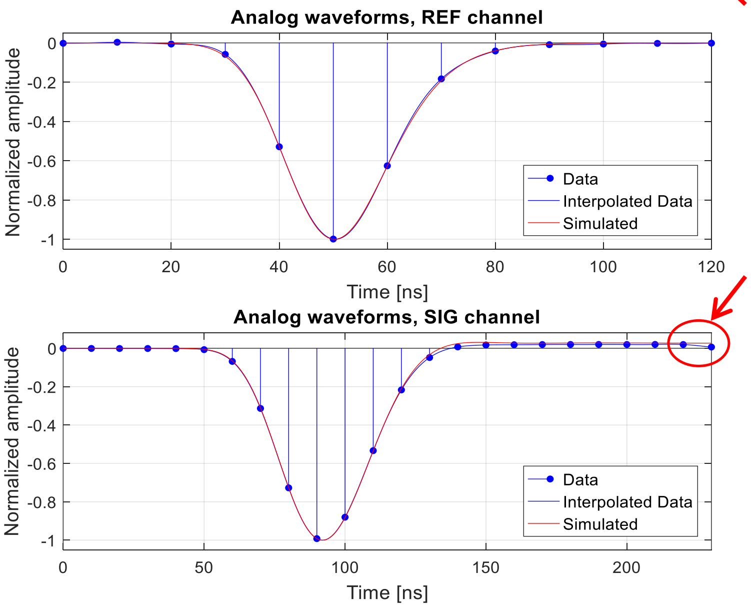
250 MHz,
CH1 = ref, CH2 = sig (15 ns)



100 MHz,
CH1 = ref, CH2 = ref

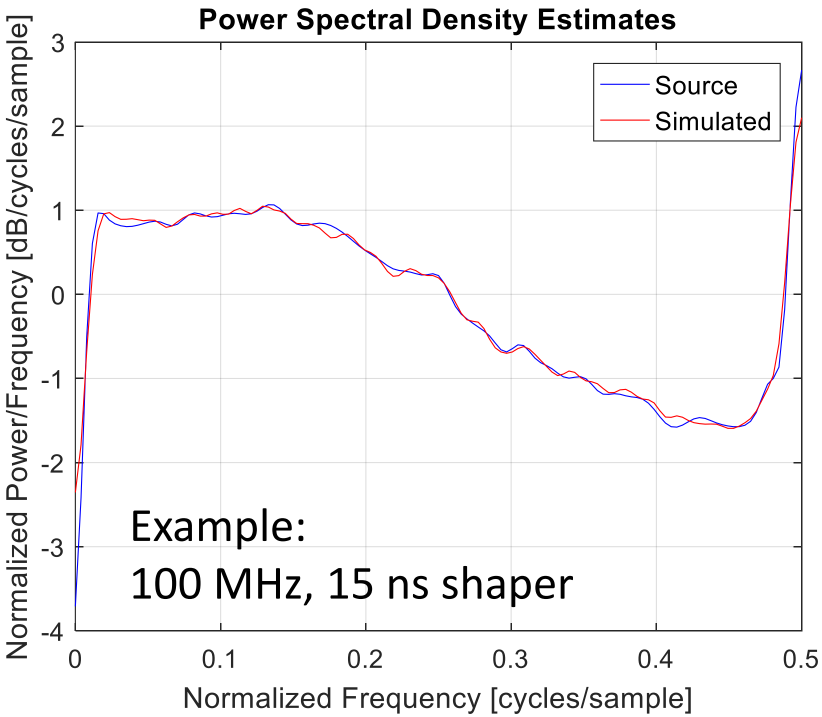


CH1 = ref,
CH2 = sig (15 ns low power)

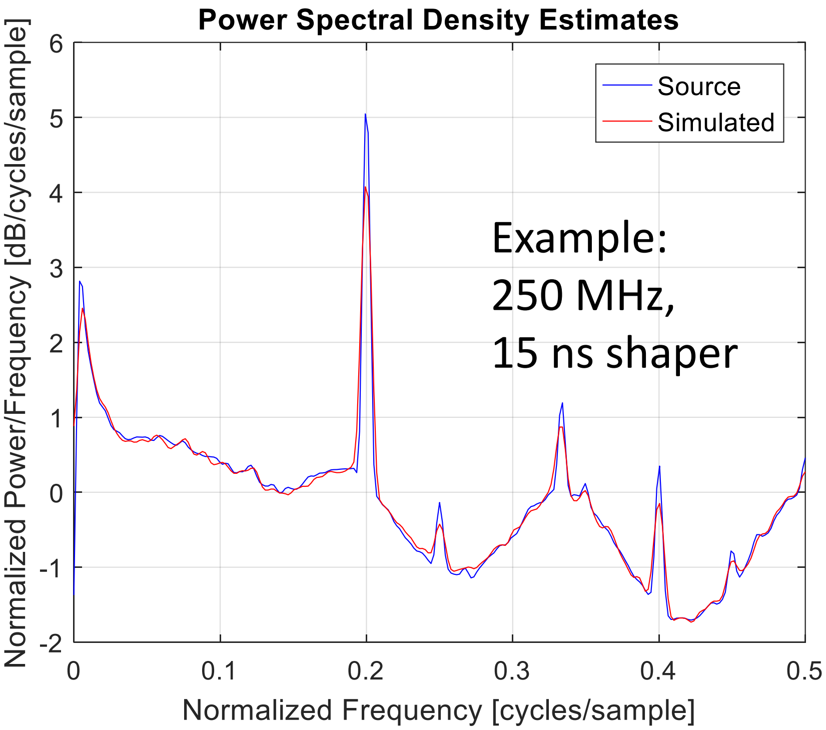


Interpolation artefacts

Noise models



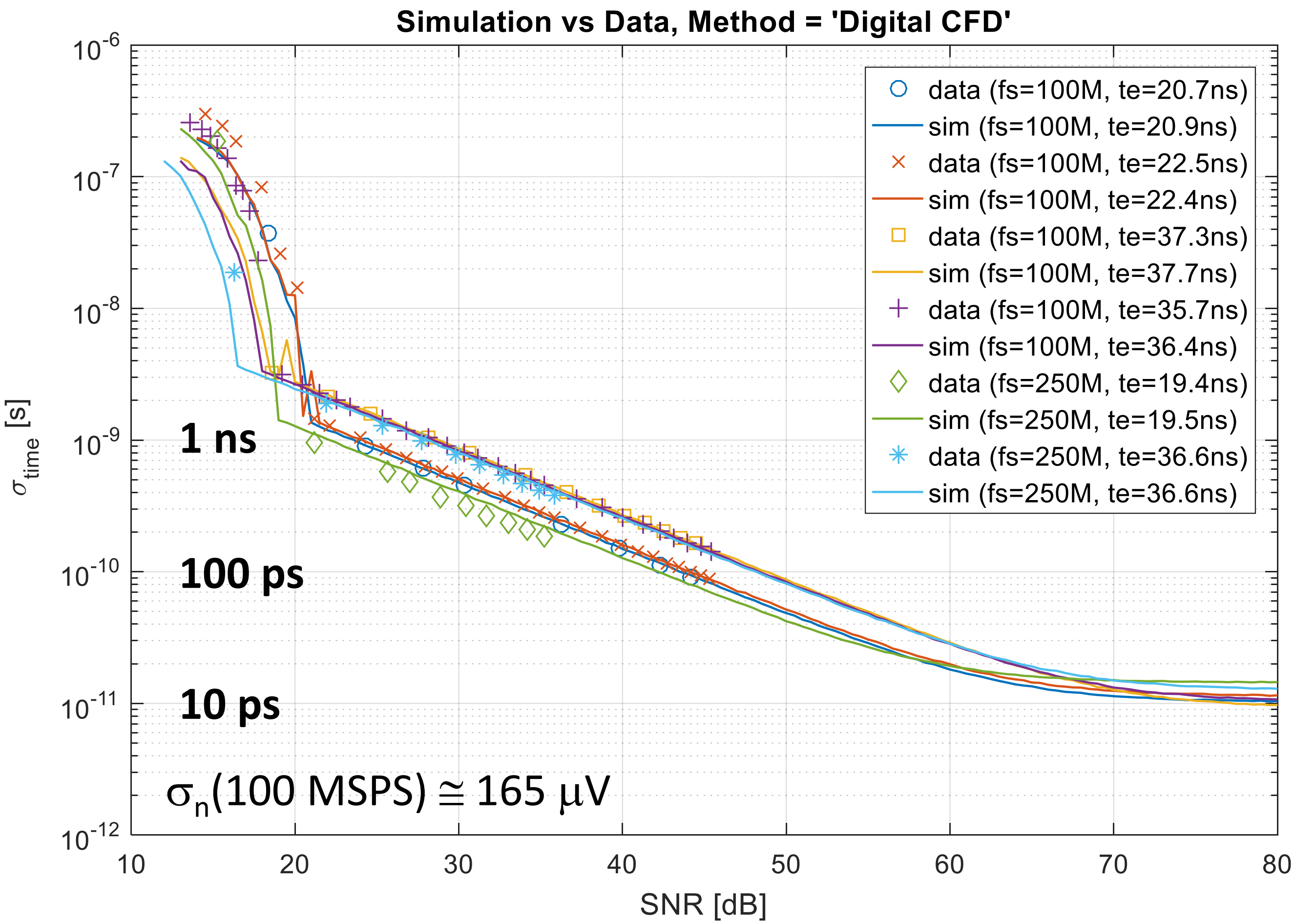
Example:
100 MHz, 15 ns shaper



Example:
250 MHz,
15 ns shaper

- Good match of simulated periodogram with an experimental one.
- Potential problem:
  - Some of the deterministic components (peaks in spectrum) do not have random phase, but are correlated to the sampling clock.

Results – Digital CFD



1 ns

100 ps

10 ps

σ_n(100 MSPS) ≅ 165 µV

mV → 0.5 1.7 5.2 16.5 52.3 165.3 523 1653

SNR ≥ 20 dB

Good match of model and data for 100 MHz ADC, slightly worse for 250 MHz ADC

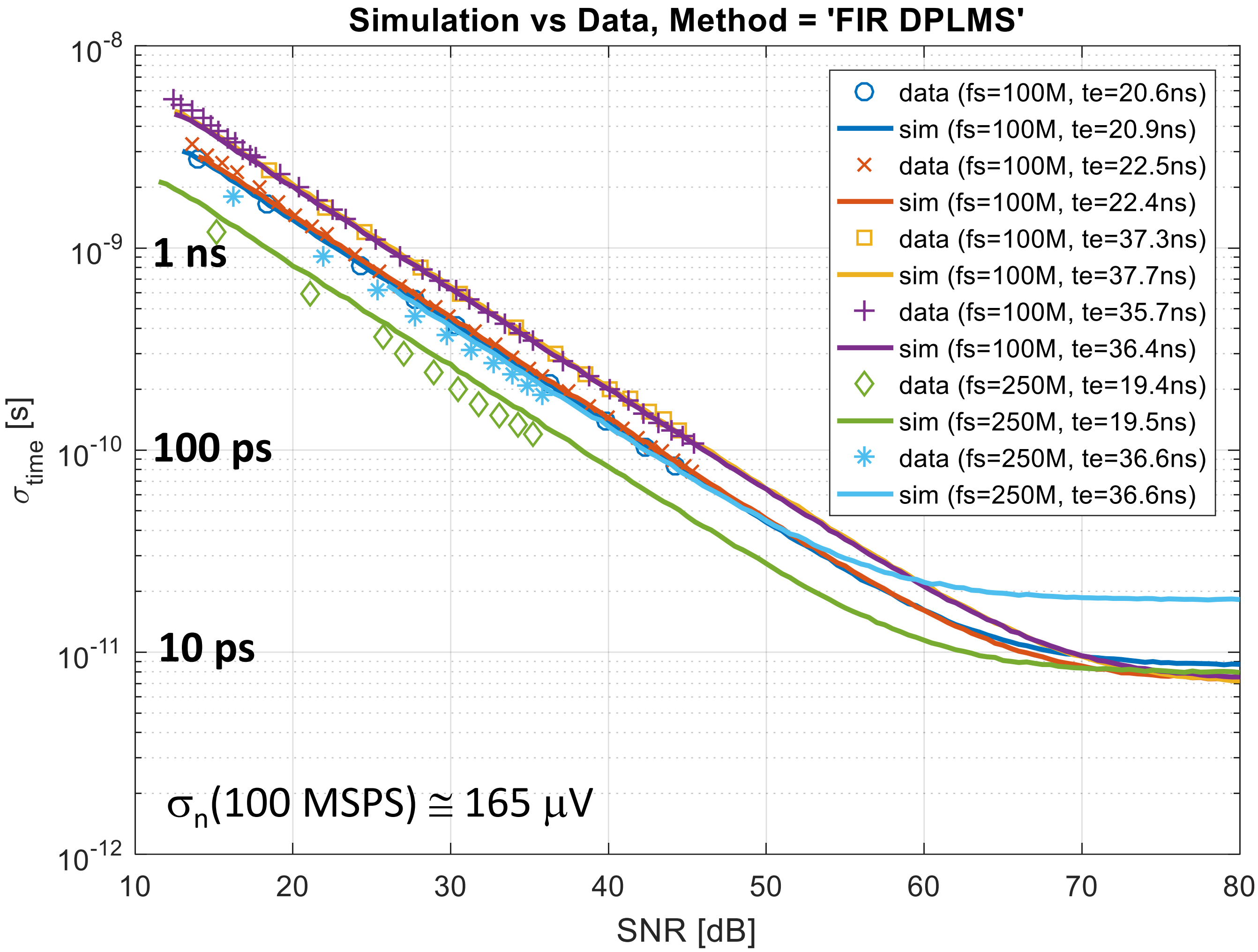
SNR < 20 dB

Poor match, data worse than model. Not a useful range anyway, as we need σ_{time} < 1 ns.

Timing resolution is proportional to

$$\frac{t_{rise}}{SNR}$$

Results – FIR DPLMS



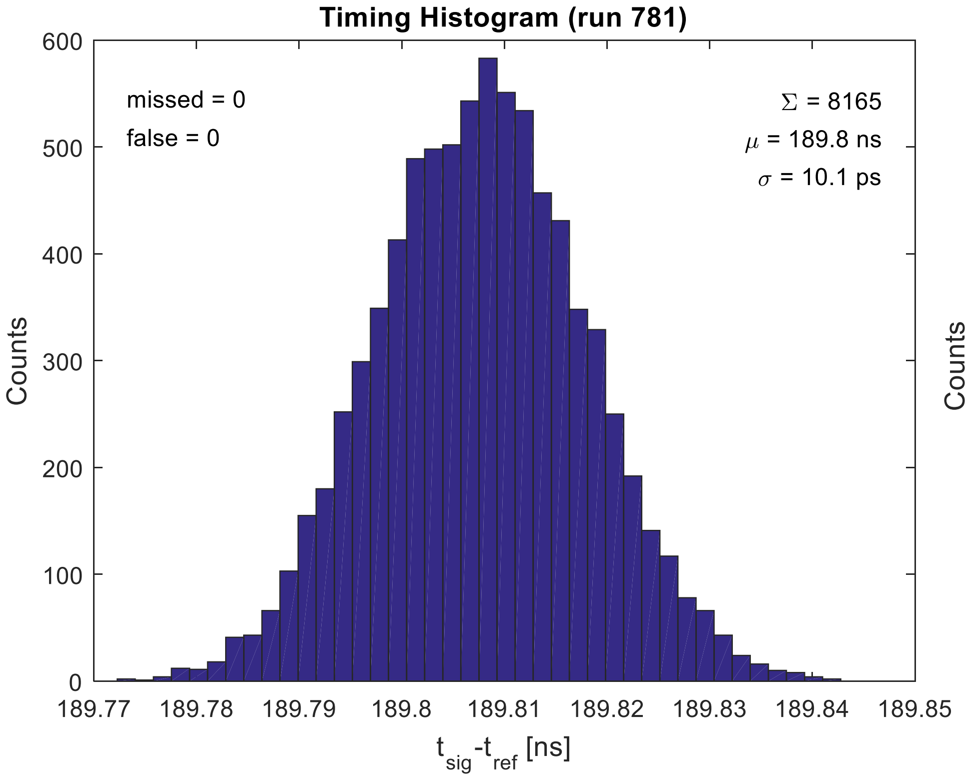
mV → 0.5 1.7 5.2 16.5 52.3 165.3 523 1653

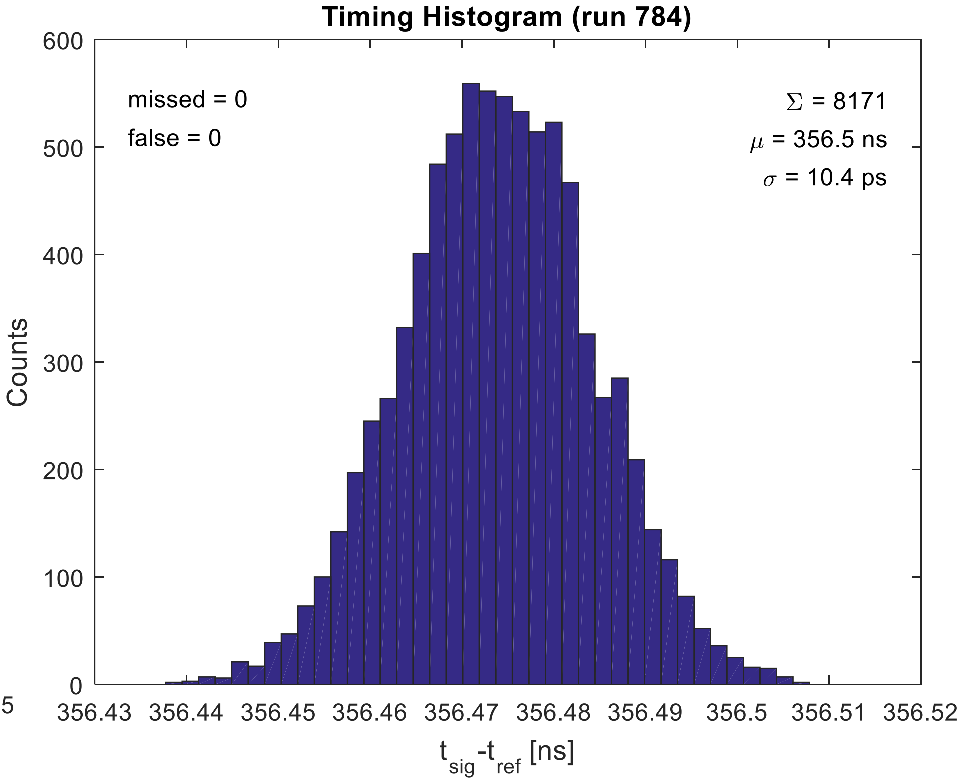
Good match of model and data for 100 MHz ADC, slightly worse for 250 MHz ADC

250 MHz data better than model – possibly due to some correlation which is not reflected by simulation.

Example Histograms – FIR Timing

Large SNR case (approx. 60 dB)

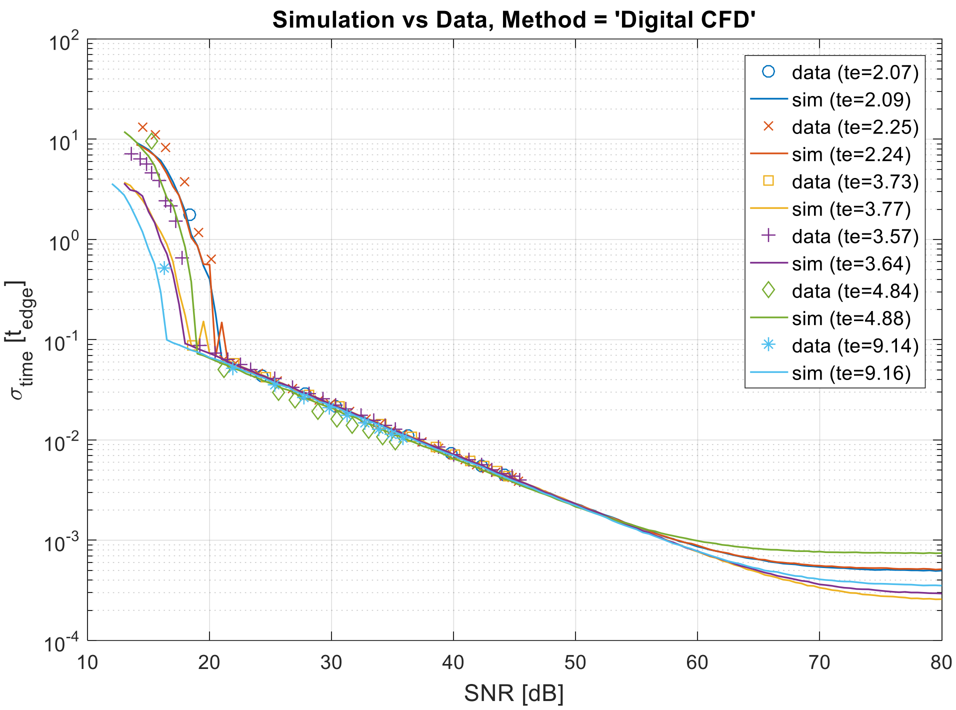


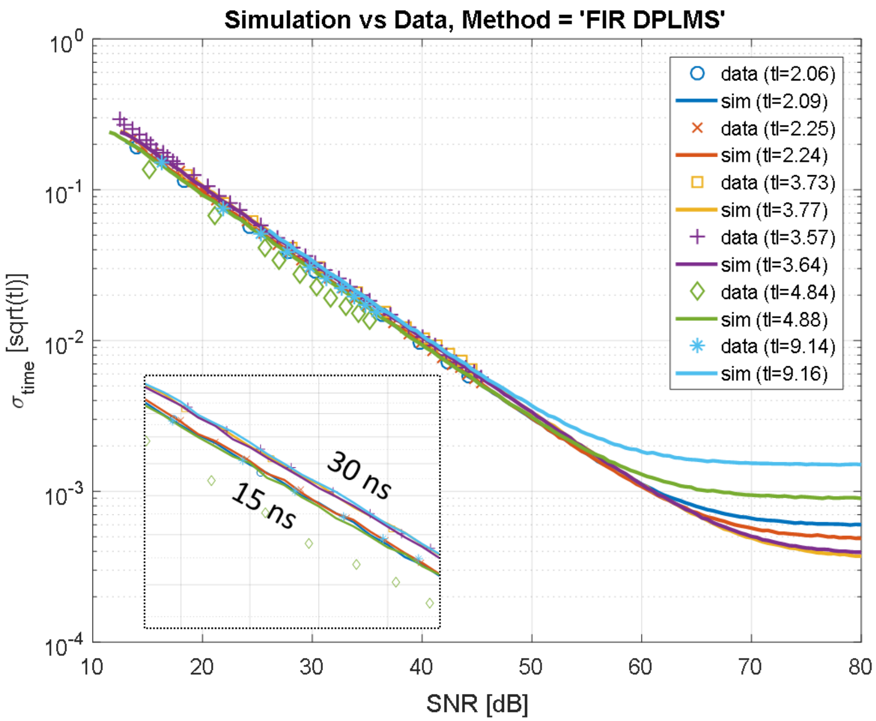


100 MSPS ADC, 14-bit, 15 ns shaper

10 ps resolution from a system with 10 ns sampling

Digital CFD / FIR DPLMS – Normalized



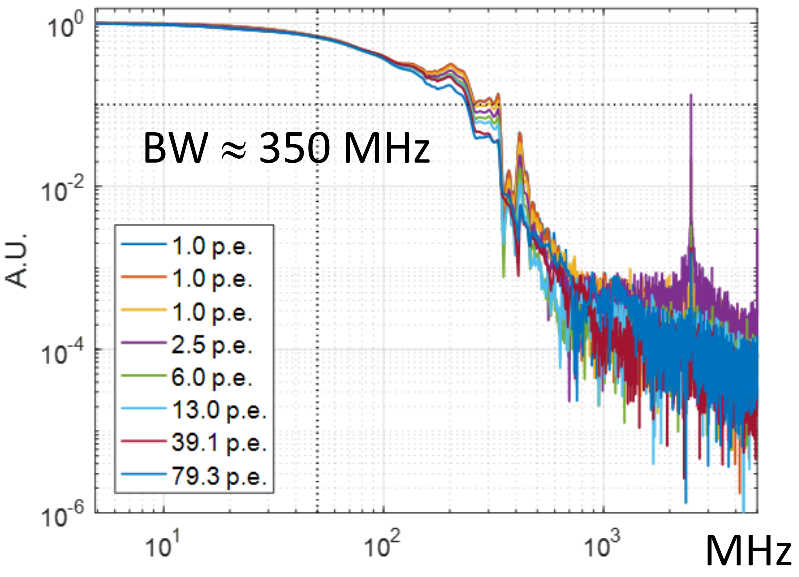


- Don’t need extremely high sampling rates to maintain good timing resolution, as long as SNR is sufficient
- It seems that it is better to maintain sharp edge → logical, as we don’t cut bandwidth of the signal that still has valid information
  - Sharp edges help in pile-up resolution
- Oversampling help only in case of FIR-based algorithms → SNR gets better

R14374 – Waveforms

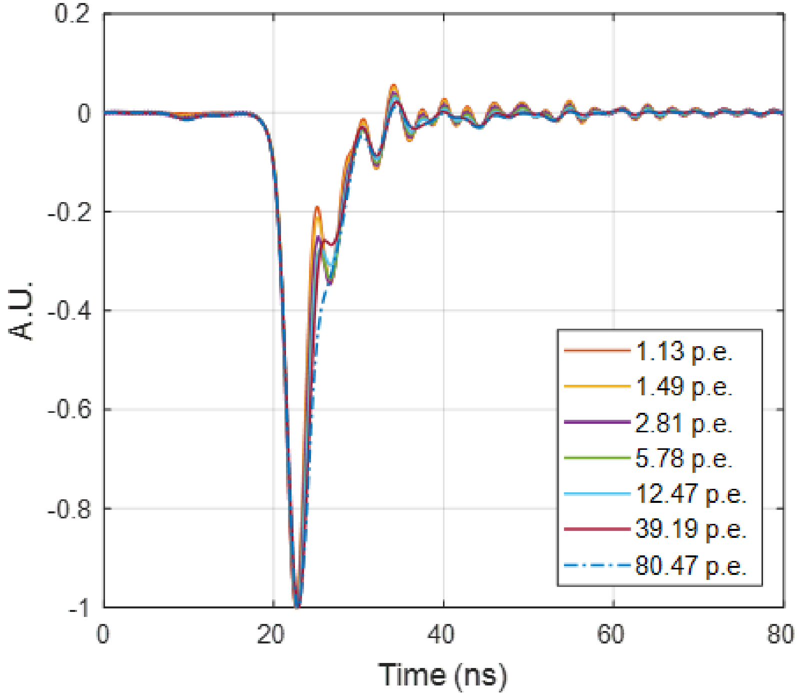
- Visible dependence of waveform shape on position of the light source on the photocathode
- $t_{rise} \in$ (1.9 ns, 3.0 ns), FWHM $\in$ (3.0 ns, 4.7 ns); both increase with PE level (expected)

Normalized Amplitude Spectrum

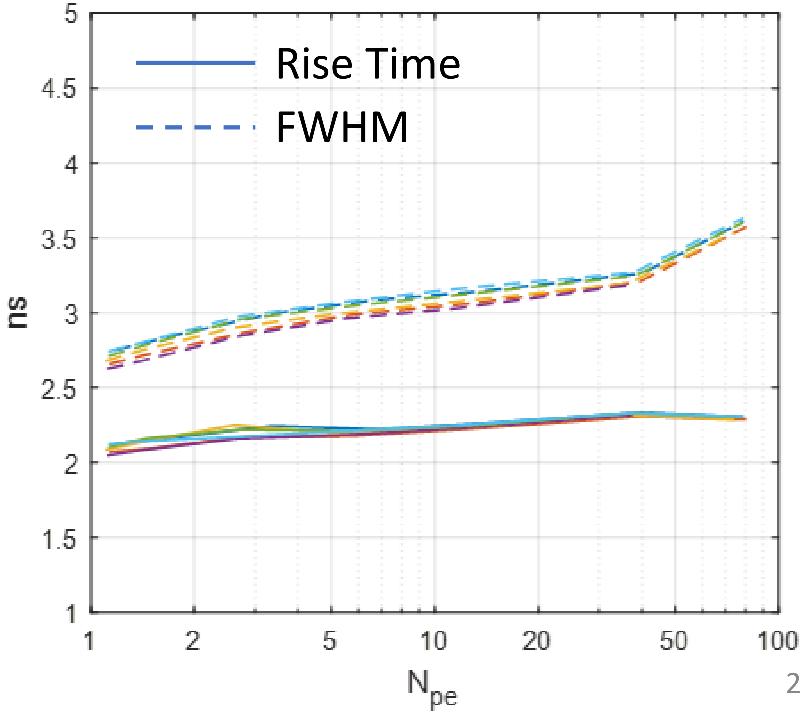


BW ≈ 350 MHz

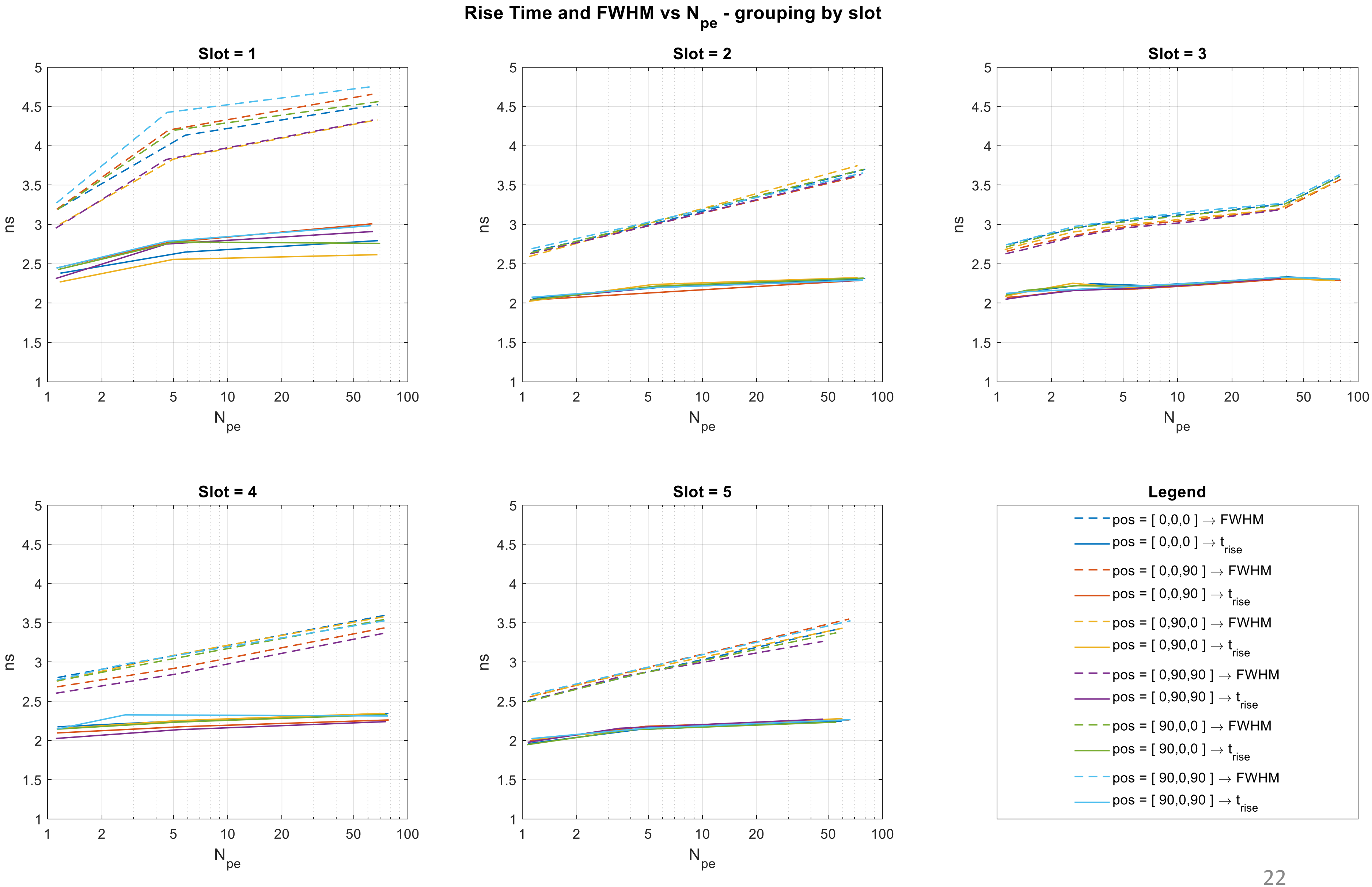
Normalized templates



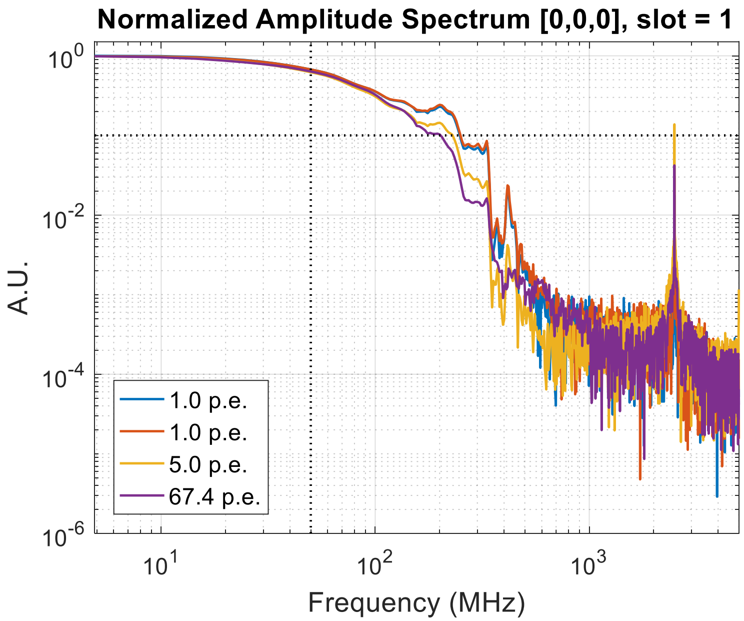
Rise Time & FWHM

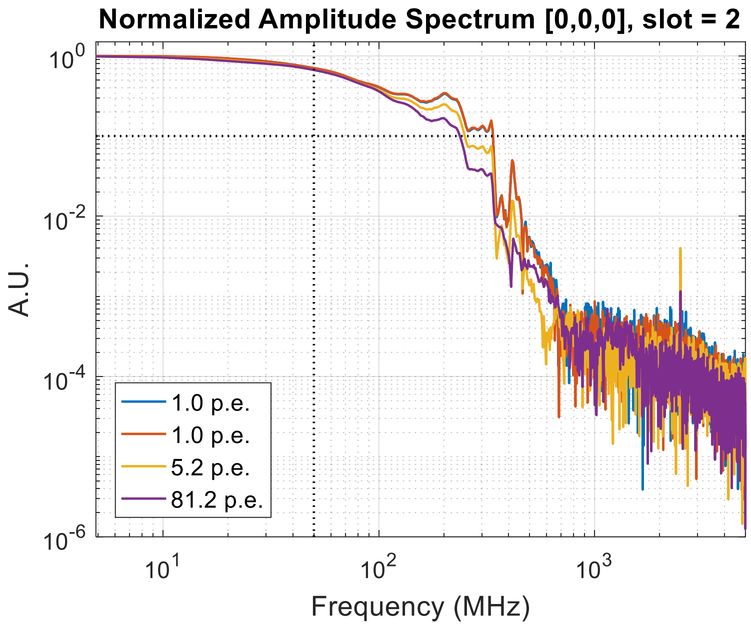


R14374 - Waveform Shape



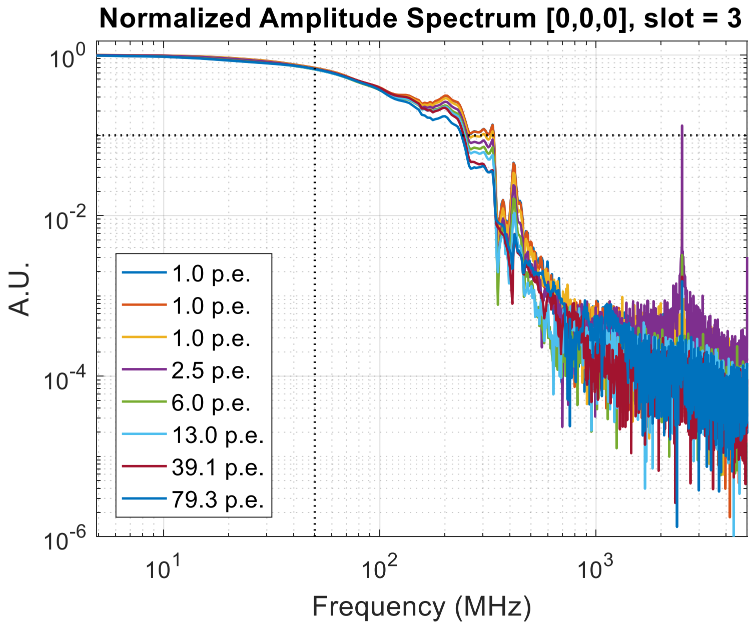
R14374 - Pulse Bandwidth

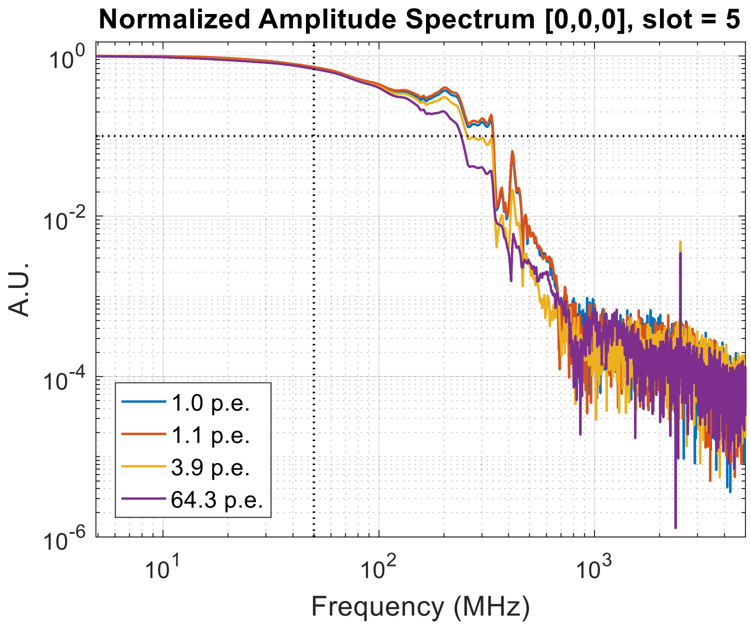

Normalized Amplitude Spectrum [0,0,0], slot = 1


Normalized Amplitude Spectrum [0,0,0], slot = 2

Bandwidth: ≈350 MHz

Bandwidth almost unchanged in the passband region of the anti-aliasing filter

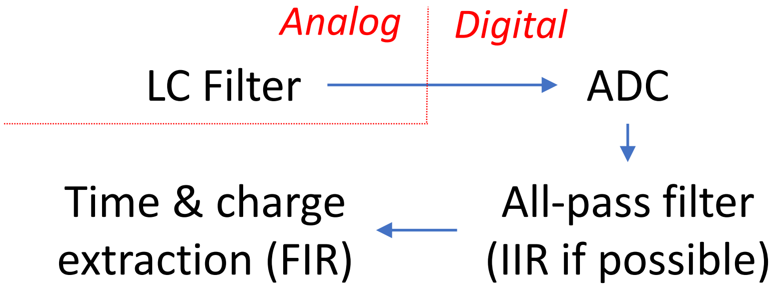

Normalized Amplitude Spectrum [0,0,0], slot = 3


Normalized Amplitude Spectrum [0,0,0], slot = 5

At sufficiently low cut-off pulse shape will be constant – but we will loose in pile-up resolution

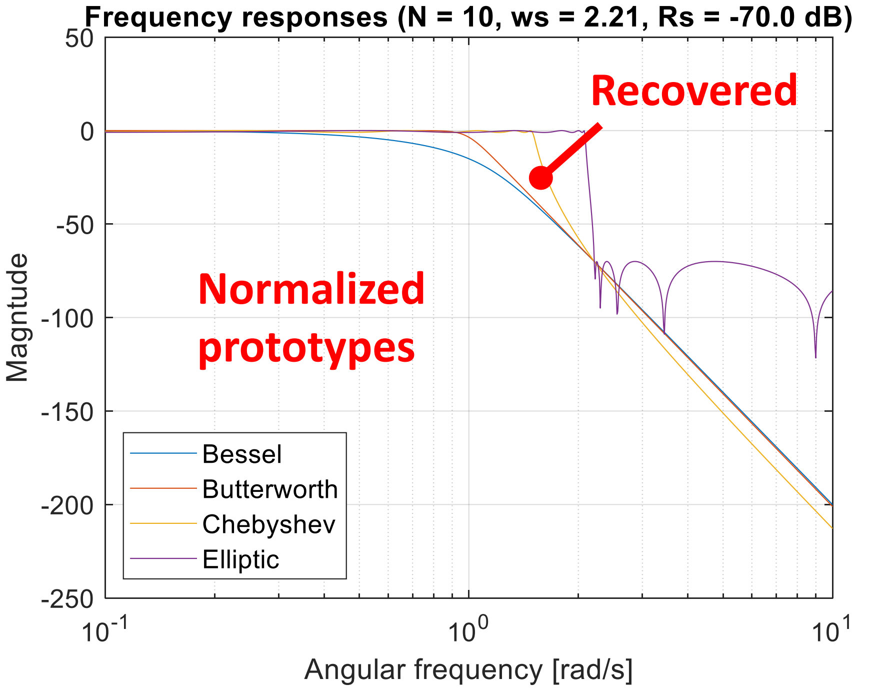
Where are we now?

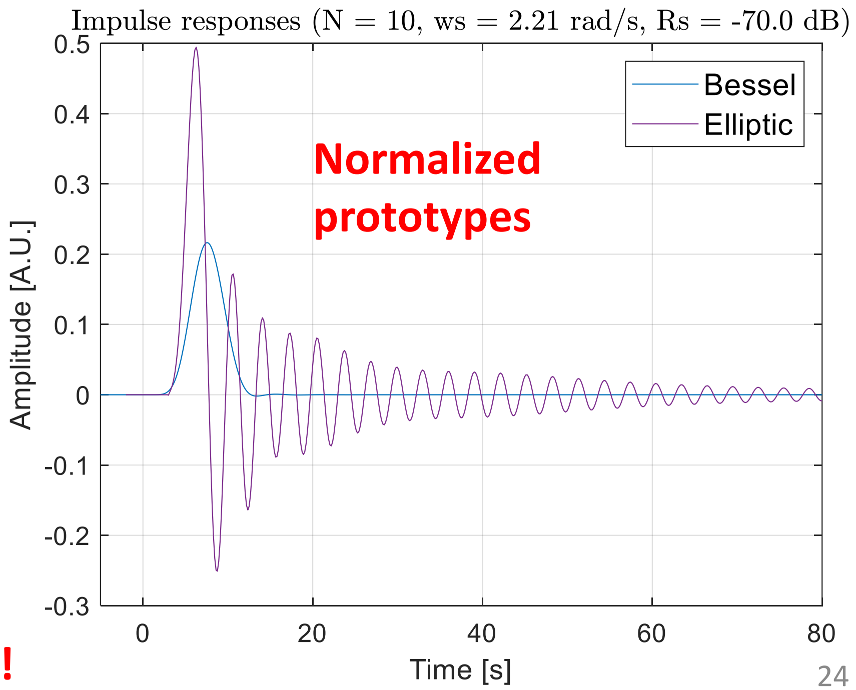
- Re-designing the shaper
  - Old shaper used for tests was too noisy, had too low cutoff frequency
  - Decided to switch to fully passive design (LC-ladder) – still need one amplifier to separate LC circuit from the twisted pair
  - Switch from Bessel to elliptic (hopefully)
- Need additional digital all-pass filter to correct passband ripple and phase



Analog
Digital
LC Filter
ADC
All-pass filter (IIR if possible)
Time & charge extraction (FIR)

Check needed if this is possible!



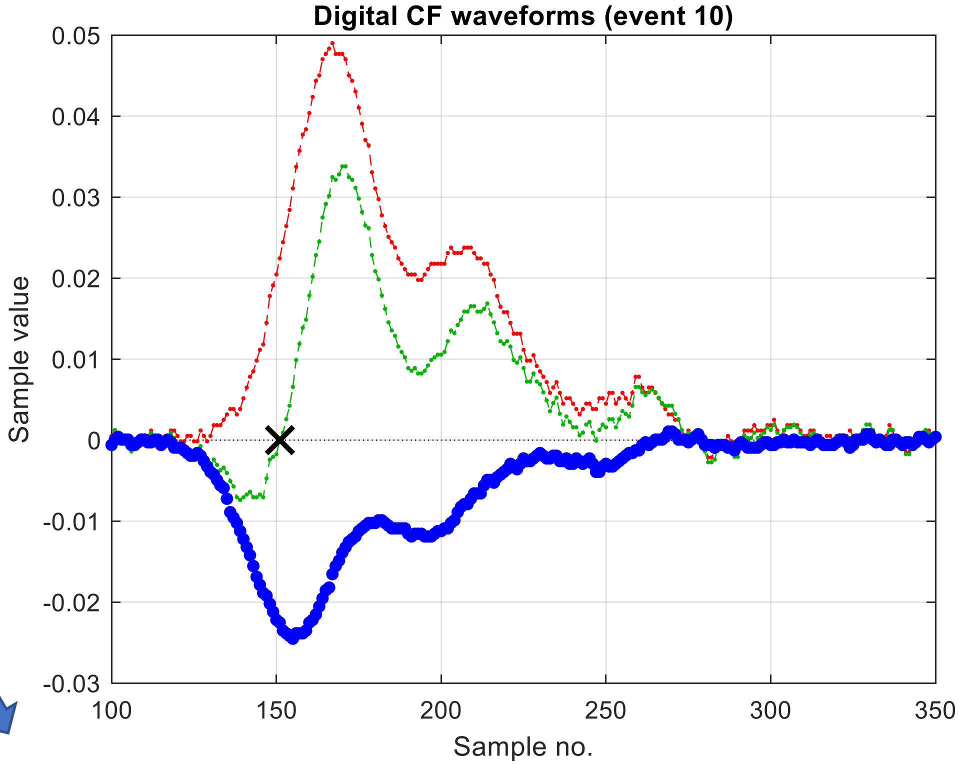


Summary

- Much work already done
- Prototype foreseen in July
- Need to foresee that in FIR-based methods the estimate may be completely wrong in case of non-standard shape (for ex. pile-up)
    - Need quality factor for each time/charge estimate
    - Should send full waveform for off-line processing
- We’re also involved in photosensor characterization
    - Can’t design good electronics without understanding signal source

Revised time estimation

- Digital CFD – limit shift to leading edge only – trailing edge not well defined
- For FIR-based method, depending on cutoff frequency we many need to parameterize impulse response of the filter wrt. charge

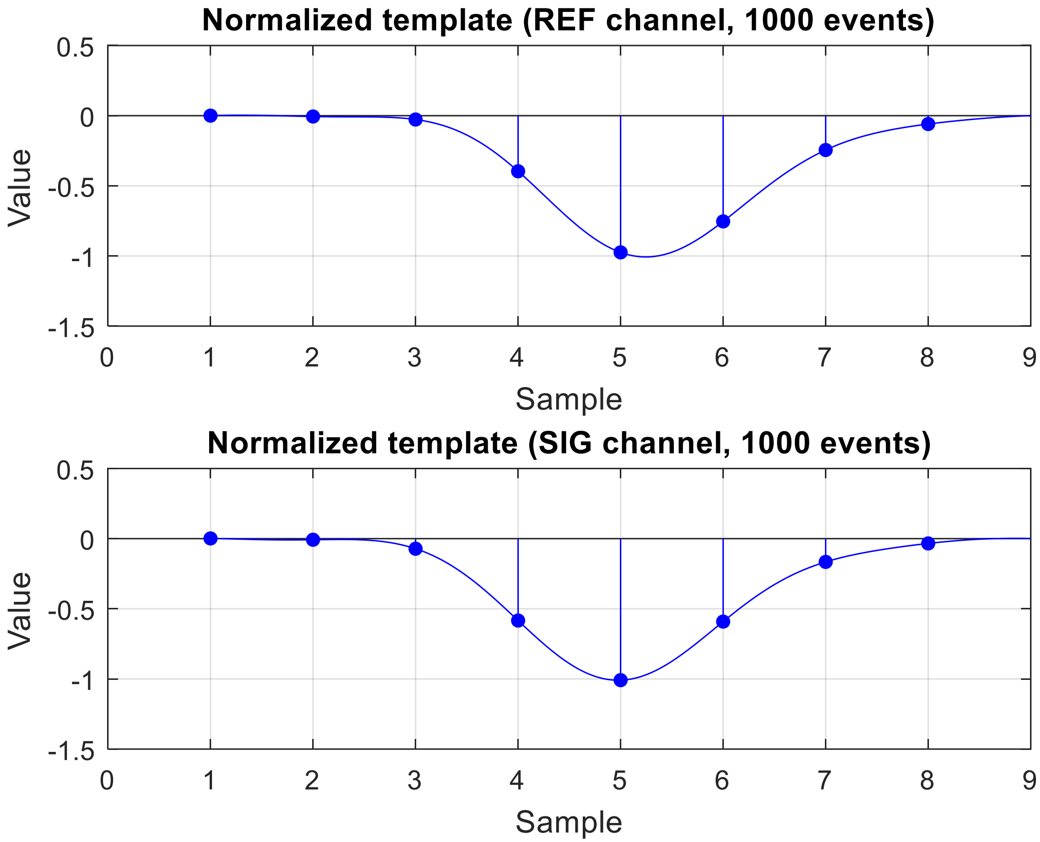


Significant increase in data rate – need efficient coding and possibly lossy waveform compression

BACKUP

FIR synthesis

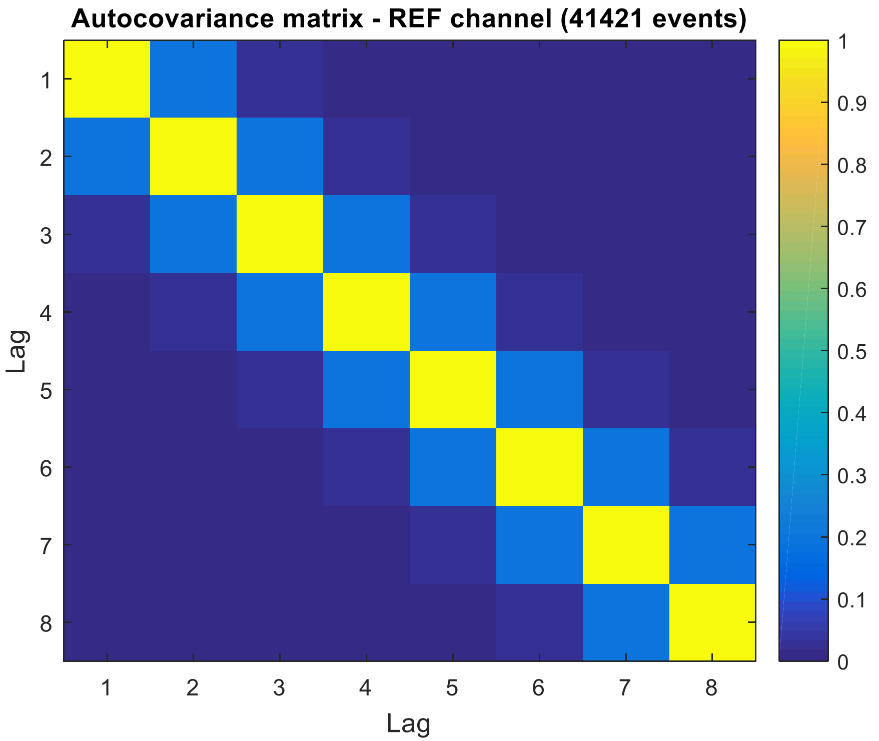
STEP 1: Detect template

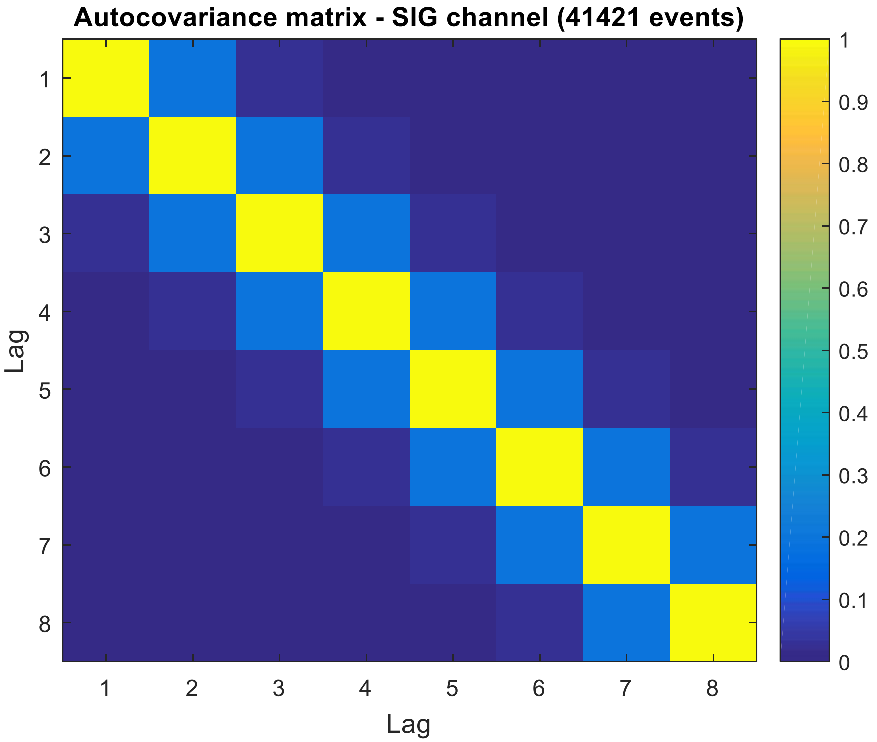


- Compute cross-correlation between two events.
- Align pulses using sinc interpolation – resample 2nd event to maximize cross-correlation.
- Average events.
- Take next event and resample it to maximize cross-correlation with the averaged event.
- Repeat last step for desired amount of events.

FIR synthesis

STEP 2: Calculate noise autocovariance matrix



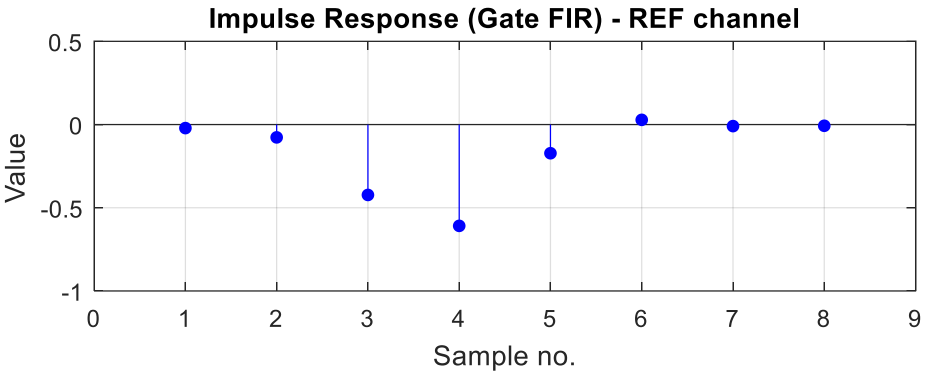


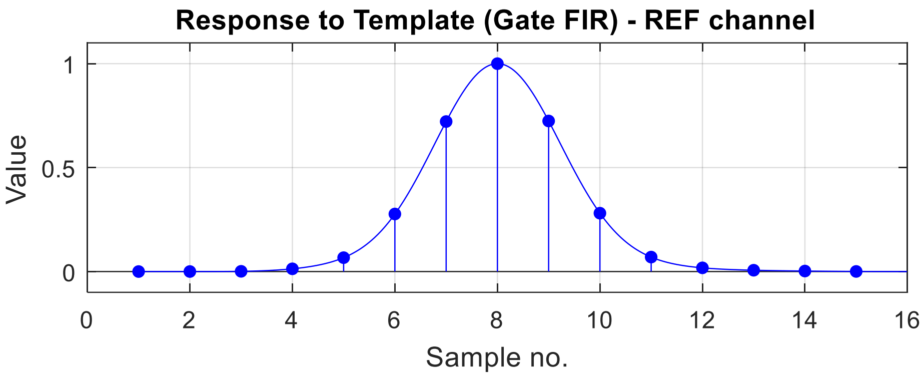
If the images are smeared, then it is PDF's image compression rather than strange covariance matrix.

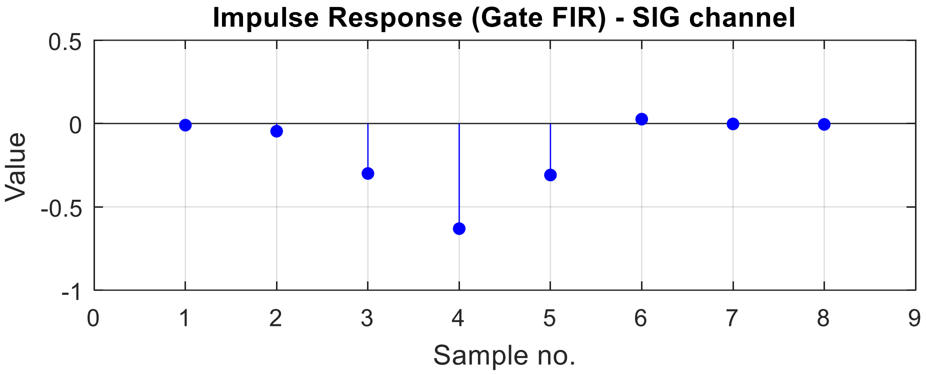
FIR synthesis

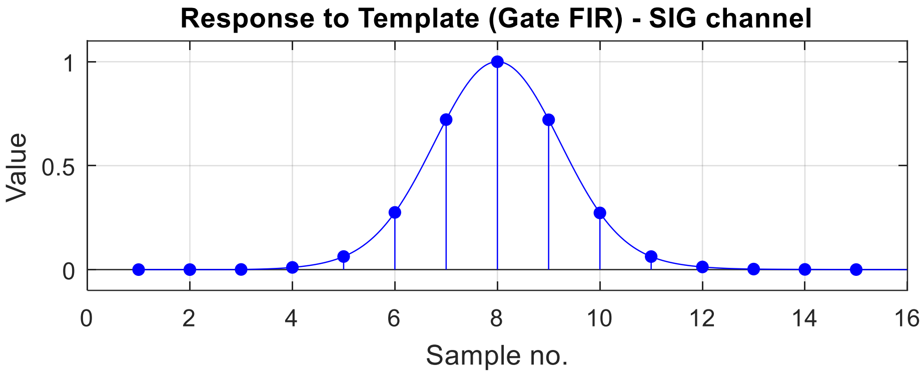
STEP 3: Calculate ‘gate’ filter

The ‘gate’ filter will be used to detect pulse. It is a standard matched filter that maximizes SNR.


Impulse Response (Gate FIR) - REF channel

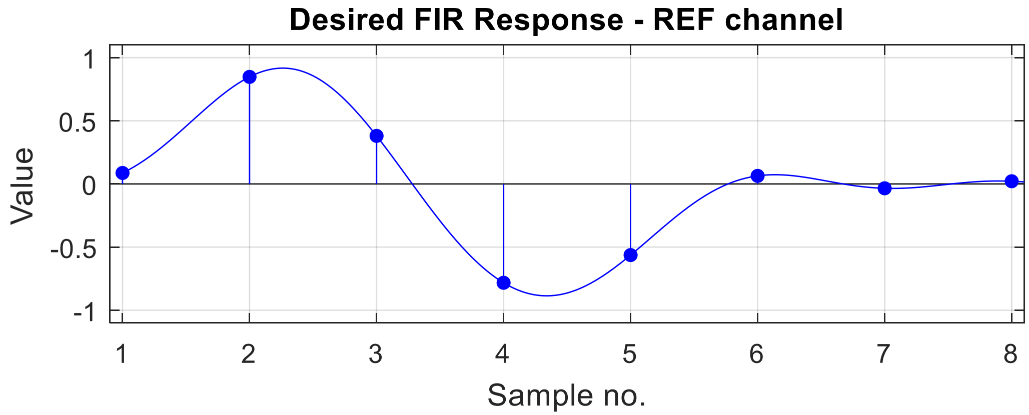

Response to Template (Gate FIR) - REF channel

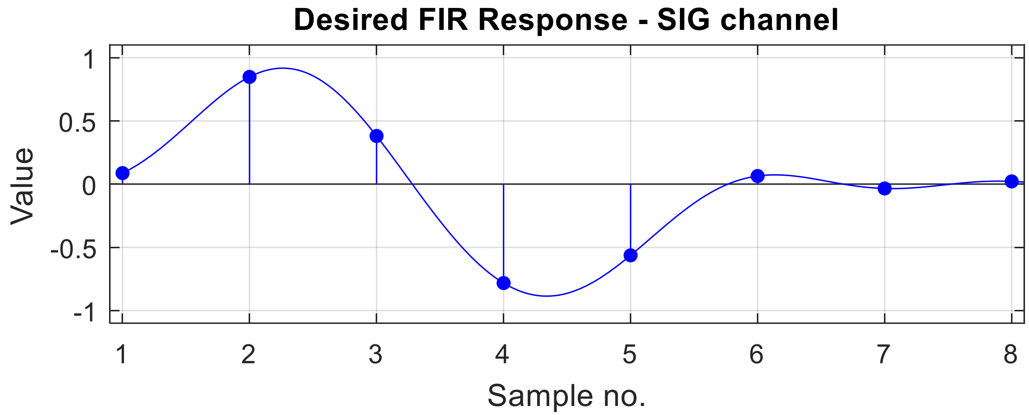

Impulse Response (Gate FIR) - SIG channel


Response to Template (Gate FIR) - SIG channel

FIR synthesis

STEP 4: Calculate desired FIR response



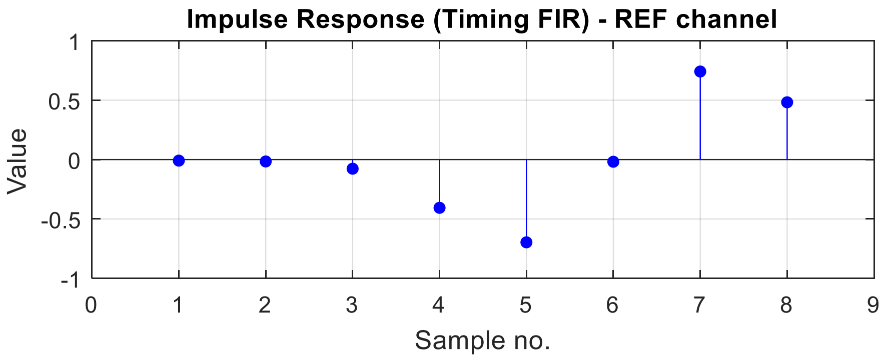


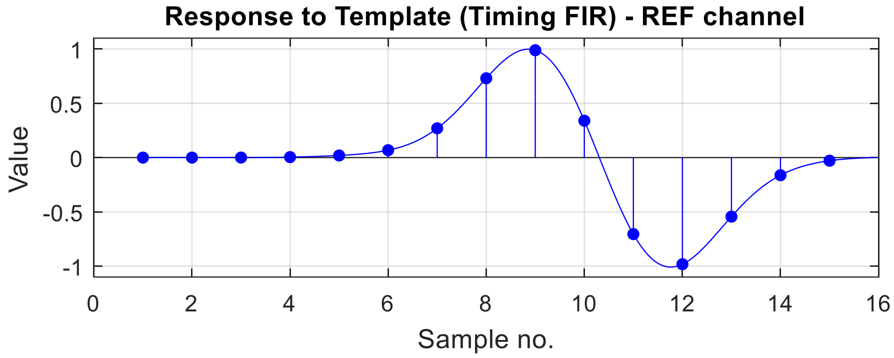
- Use solver and compute waveform shape that meets desired shape, length and linear edge requirements.
- Downsample resulting waveform so that Nyquist criteria is met.
- Figures show downsampled responses.

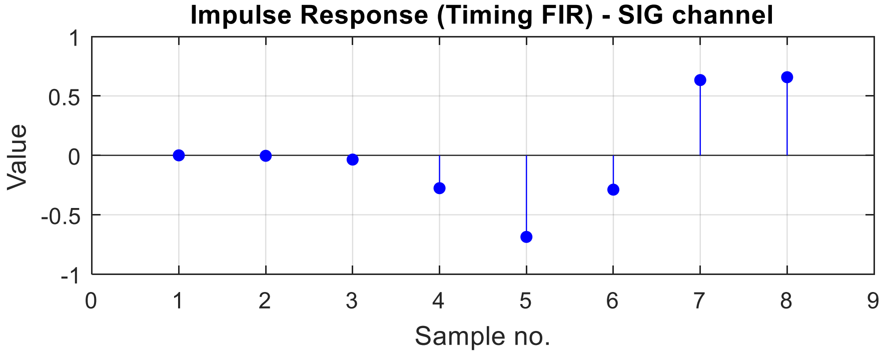
FIR synthesis

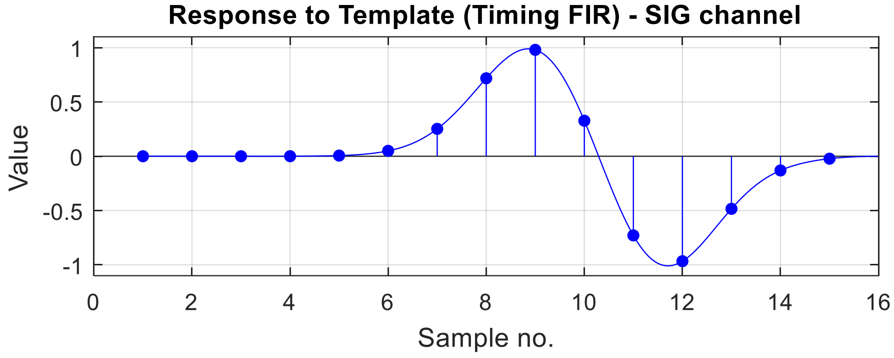
STEP 5: Calculate ‘timing’ FIR

- Use DPLMS method to calculate FIR filter based on pulse template, desired response and noise autocovariance matrix.


Impulse Response (Timing FIR) - REF channel

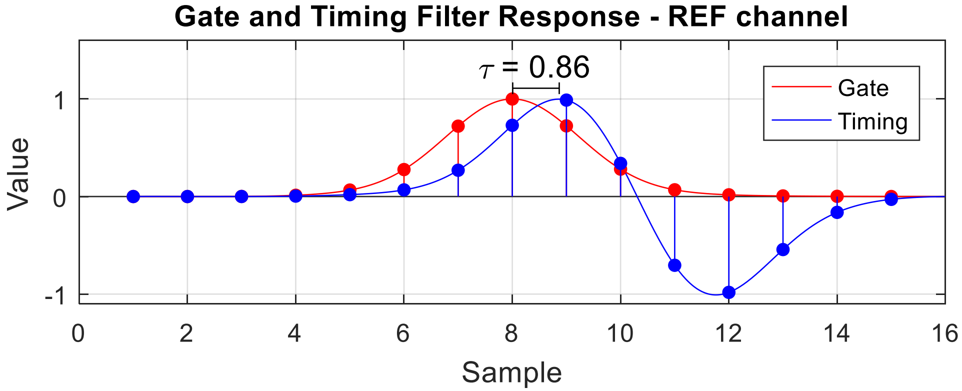

Response to Template (Timing FIR) - REF channel

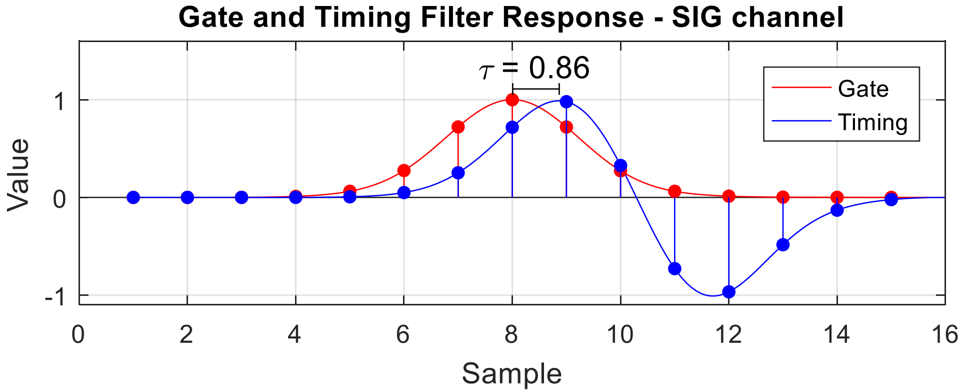

Impulse Response (Timing FIR) - SIG channel


Response to Template (Timing FIR) - SIG channel

FIR synthesis

STEP 6: Calculate shift between maximums of ‘gate’ and ‘timing’ filter response

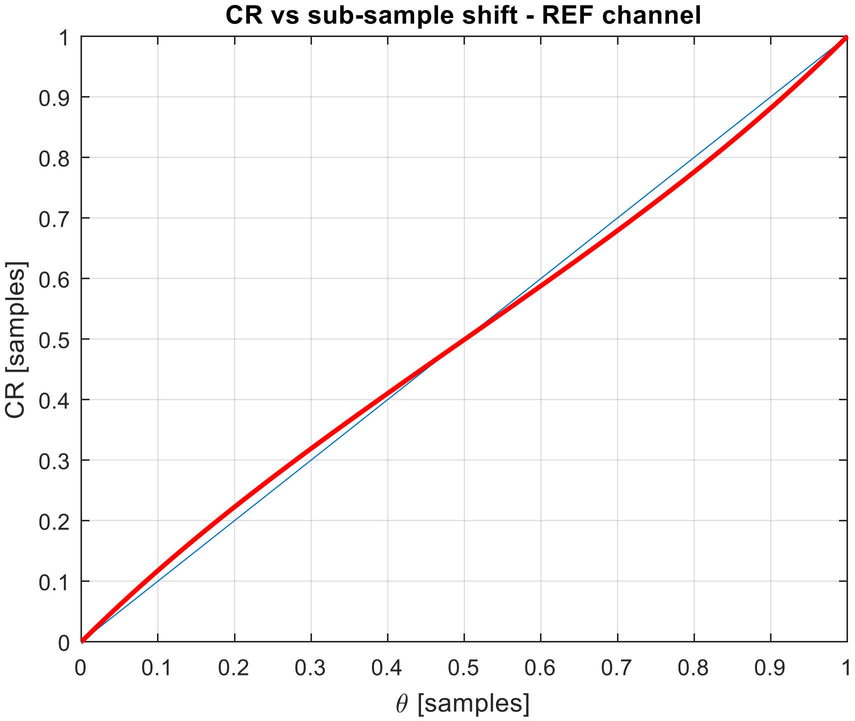




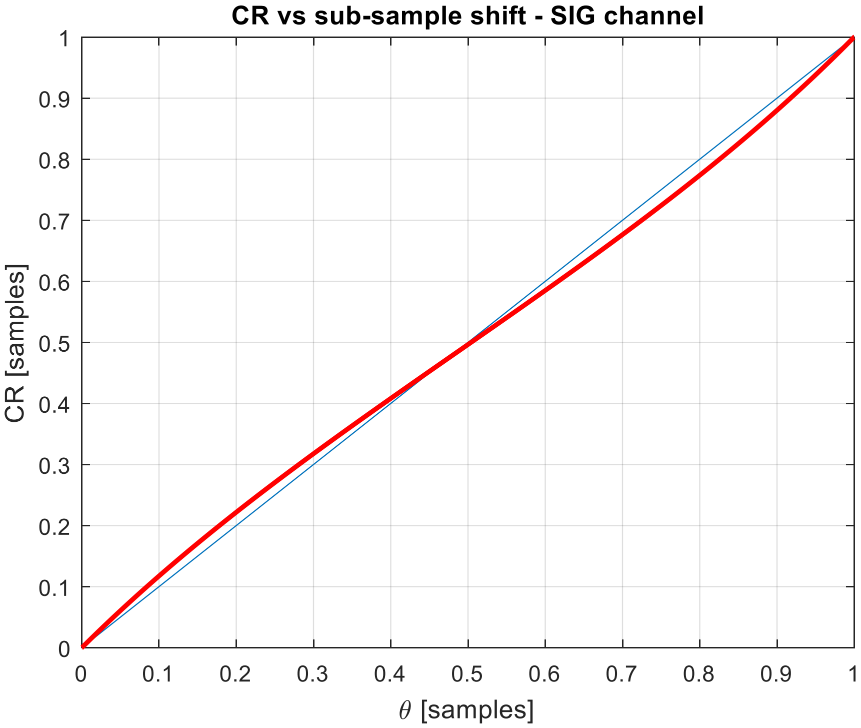
- Make separate calculation for ‘reference’ and ‘signal’ channels
- This value will later be used to start searching for zero-crossing of ‘timing’ filter response.

FIR synthesis

STEP 6: Calculate correction function to account for non-linear shape near zero crossing of 'timing' filter response



CR vs sub-sample shift - REF channel



CR vs sub-sample shift - SIG channel

θ - actual sub-sample shift

$$CR = \frac{P}{P - Q}$$